A MODEM FOR PACKET RADIO COMMUNICATION

by

CAPT. P. K. MALHOTRA

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DEPARTMENT OF ELECTRICAL ENGINEERING
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CERTIFICATE

Certified that this work 'A MODEM FOR PACKET RADIO COMMUNICATION' by Capt. P.K. Malhotra has been carried out under my supervision and has not been submitted elsewhere for a degree.

May, 1985

K R Srivathsan

(K.R. Srivathsan)
Assistant Professor
Department of Electrical Engineering
Indian Institute of Technology
KANPUR

POST GRADUATE OFFICE
This thesis has been approved
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ABSTRACT

Packet radio is a technique that extends the original packet switching concepts to the domain of broadcast radio networks. All users in a packet radio are assumed to share a common radio channel, access to which is controlled by microprocessors using appropriate protocols. A key component of the packet radio system is the baseband signal design and carrier modulation suited to fast timing recovery and acceptable data rate over limited channel bandwidth.

In this thesis modified duobinary signalling has been used for spectral reshaping of the baseband digital signal, which is then transmitted over FM radio channel. Modified duobinary has been specially chosen because it has no DC component and provides builtin partial error detection. Various protocols which can be used for packet radio environment have also been reviewed.

A tree based multiaccess protocol has been suggested for implementation in a packet radio.

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CHAPTER 1

LOCAL AREA NETWORK AND PACKET RADIO

1.1 INTRODUCTION:

The first computer network system to employ radio for its communication facility, is the ALOHA system [1,2] at the University of Hawaii, which is spread out on four islands. In this system a small FM radio transmitter/ receiver with sufficient range (30 km) was preferred over telephone lines which were expensive and unreliable. In future, as the electronic devices multiply throughout the home and office, low power packet radio would permit all these devices to communicate among themselves and with similar devices throughout the world via a master station tied into a public data network. In this thesis an attempt to design and develop a radio link suitable for ground packet radio has been made. A review of various protocols applicable to packet radio has also been done.

1.2 Review of Packet Radio:

In 1970, ABRAMSON and his colleagues at the University of Hawaii developed ALOHA system, using ground based radio packet broad-casting, to access the main

computer centre. It consisted of a set of terminals linked directly to the computer centre by UHF packet radio. A schematic diagram of original ALOHA system [3] is given in Fig. 1.1. In this system all terminals were in the line of sight of the antenna, atop the computer centre which served as the hub or station for all communications. For such data communication broadcast radio has been chosen as an affective alternative over line communication due to the following reasons [4].

a) In a broadcast mode any number of users may access the channel.

b) A broadcast mode is particularly suitable when the users are mobile or are located in remote regions where a wire connection is not easy to implement.

c) The design of broadcast system is flexible e.g. the packet radio communication system became operational with two or three users. Size can be increased upto channel capacity.

d) It can serve a large population of active/inactive users.

e) Line communication is expensive

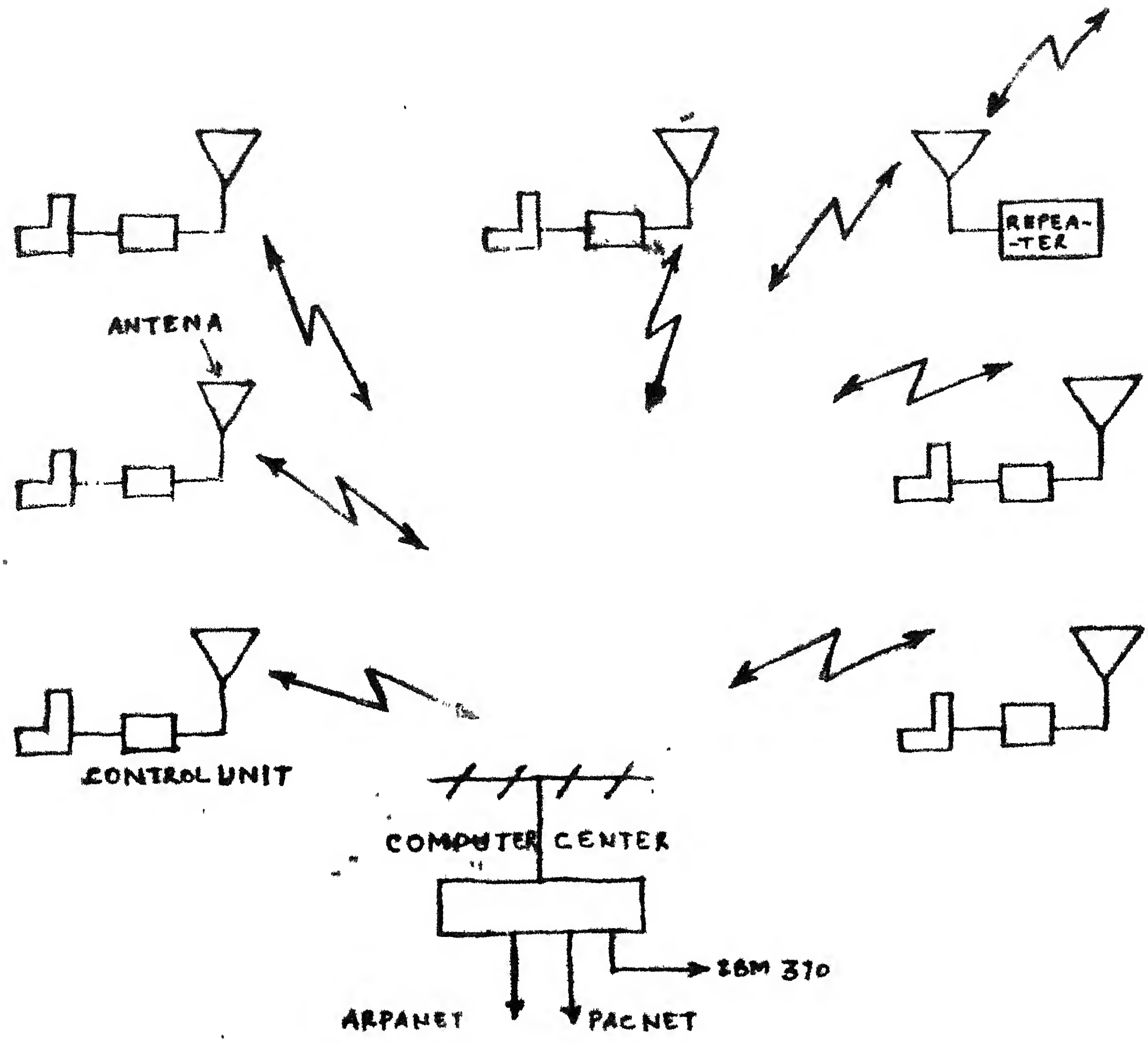


Fig. 1.1 The University of Hawaii ALOHA System

f) Multiaccess Broadcast Capability: This capability in radio communication may be useful for certain multipoint to multipoint communication application.

In the ALOHA system when a station has data to send, it just goes ahead and sends. This is called PURE ALOHA. When the central computer receives a packet, it inserts an acknowledgement packet into the output stream. If a station does not receive an acknowledgement within a preset time, it assumes that the packet suffered a collision and retransmits it. In this the maximum channel utilization is 18%. In 1972, ROBERTS published a method of doubling the capacity of an ALOHA system and called it SLOTTED ALOHA [6] (explained in Chapter 2). KAHN in his paper [5] in 1977 has suggested the extension of Hawaii network to the use of repeaters to achieve area coverage beyond line of sight. A unified presentation of packet broadcasting theory is given by ABRAMSON [6]. In this paper, performance of packet broadcasting, when users of network have variety of data rate is also given. BINDER and JACOB [10] have described the use of packet radio for general purpose packet satellite networks. Using a satellite channel, we can implement a packet communication network with a large population of geographically distributed users.

One of the most important problems in the design of such a network is, how the users can effectively share a single satellite channel[11]. Many multiaccess protocols (SALOHA, RALOHA, TDMA Reservation, SRUC) have been proposed for this and also performance comparison is given in SLAM'S paper [12,13]. The inherent long propagation delay of 0.25 second for a single hop satellite communication is long compared to transmission time of a packet. This has major impact on bandwidth allocation and on the error and flow control protocols. Detailed review of the various protocols for packet radio is given in Chapter 2.

1.4 OUTLINE OF THESIS:

In this thesis an attempt to study the packet radio communication and its implementation for a geographically distributed (over a few km) computer users like IIT Kanpur has been made. In this chapter the various packet radio systems have been briefly discussed. In Chapter 2 the design considerations and various protocols for packet radio are reviewed. In Chapter 3 we study modified duo-binary signalling [10] scheme and how it can be used in conjunction with ground radio system for achieving packet communication. For our purpose we assume that a UHF radio set for transmitting and receiving the MDB is available. In

Chapter 4 the actual implementation of Modified Duobinary is given. Then a scheme to recover clock out of MDB signal and synchronisation is discussed.

In Chapter 5 we conclude with some suggestions for further scope of work in this topic.

CHAPTER 2

PROTOCOLS FOR PACKET RADIO

2.1 DATA NETWORK ARCHITECTURE:

To reduce the design complexity, most of the Data Networks are organized as a series of layers [3] or levels, each one built upon its predecessor. The purpose of each layer is to offer certain services to the higher layers. Between each pair of adjacent layers there is an interface. The interface defines which primitive operations and services the lower layer offers to the upper one. At the lowest layer there is physical communication between the users as opposed to the virtual communication used by higher layers. The reference model of open systems interconnection, as International Standards Organisation calls it, has seven layers (Fig. 2.1), they are:

a) The physical layer:

The physical layer is concerned with transmitting raw bits, over a communication channel. The various design issues here are, how many microseconds a bit occupies, whether transmission may proceed simultaneously in both directions, how the initial connection is established and how it is torn down when both sides are

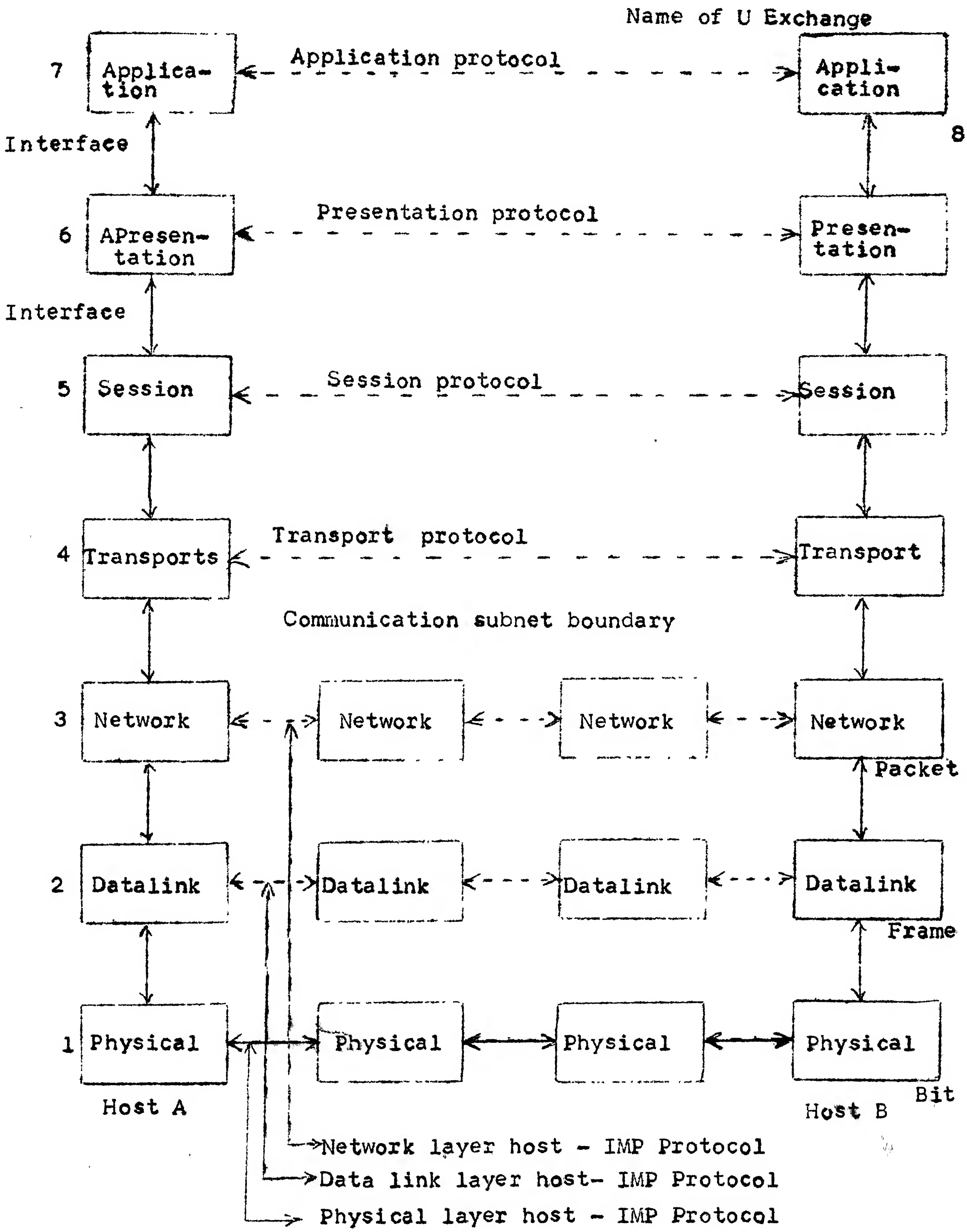


Fig. 2.1: Network Structure: ISO - OSI Reference Model

finished, how many pins the network connector has and what each pin is used for.

b) The Data link layer:

The task of the data link layer is to take a raw transmission facility and transform it into a line that appears free of transmission errors to the network layer. It accomplishes this task, by breaking the input data into data frames, transmitting the frames sequentially and processing the acknowledgement frames sent back by the receiver.

c) The Network layer:

The network layer controls the operation of the subnet. Among other things it determines how packets, the units of information exchanged in layer 3 are routed within the subnet. The communication subset which converts various hosts is called subnet.

d) The Transport layer:

The basic function of this layer is to accept data from the session layer, split it up into smaller units, pass these to network layer and ensure that the pieces all arrive correctly at the other end.

e) The Session layer:

A connection between users is usually called a session. The session layer is the user's interface into the network. A session might be used to allow a user to log into a remote time sharing system or to transfer a file between two machines.

f) The Presentation layer:

The presentation layer performs functions that are requested sufficiently often to warrant finding a general solution for them, rather than letting each user solve the problems. These functions can often be performed by library routines called by the user.

g) The Application layer:

The content of the application layer is up to the individual user. When the two user programs on different machines communicate, they alone determine the set of allowed messages and the action taken upon receipt of each.

Our aim in this chapter is to review the various Data Link level protocols as applicable to packet radio. Protocols [14] are common tools designed for controlling information transfer in a computer network. They are made up of sequences of messages with specific formats and

meanings. These messages are equivalent to the instruction of a programming language. We review various multiaccess protocols [8] in the succeeding paras.

2.2 FIXED ASSIGNMENT TECHNIQUES:

Fixed assignment techniques [8] consist of allocating the channel to the user, independently of their activity, by partitioning the time bandwidth space into slots which are assigned in a static predetermined fashion. These techniques take two common forms, Time division multiple access (TDMA) and Frequency division multiple access (FDMA). In these schemes each terminal is assigned a subchannel derived from the original channel. Such schemes avoid any collisions but are inefficient for two reasons. Firstly because terminals tend to be bursty source and therefore much of permanent assigned capacity is wasted and secondly, response time will be far worse due to scalling effect. A comparison of TDMA and FDMA is given below.

FDMA	TDMA
1. Guard Band: It wastes a fraction of the Bandwidth to achieve adequate frequency seperation.	1. Guard bands of less than 200 ns are achievable.
2. Flexibility: lack of flexibility in performing changes in allocation of bandwidth.	2. It is more flexible than FDMA

3. Complexity: It is less complex than TDMA as it requires only frequency synchronisation.	3. TDMA is more complex to implement as it requires frequency , phase bit timing and frame synchronisation for each burst.
4. Connectivity: Connectivity is not very good.	4. Connectivity is better as all receivers listen to same channel while sender transmit on same channel at different time.
5. Packet delay: large packet delay	5. Small packet delay

A dynamic control scheme such as reservation TDMA makes use of a reservation subcarrier through which terminals place request for reserved space on the data channel. This system permits dynamic allocation of channel capacity according to a terminal's demand but requires overhead in order to set up these reservations.

2.3 RANDOM ACCESS TECHNIQUES:

In data communication, the user requires the communication resources infrequently but when he does, he requires a rapid response. That is, there is an inherently large peak to average ratio in the required data transmission rate. If fixed subchannel allocation schemes are used, then one must assign enough capacity to each subscriber to meet his peak transmission rates with the consequence that

the resulting channel utilization is low. A more advantageous approach is to provide single sharable channel to the large number of users. The strong law of large numbers then guarantees that with a very high probability, the demand at any instant will be approximately equal to the sum of the average demands of that population. Packet radio is a natural means of sharing a channel. The sharing of channel has given rise to various random access protocols

a) Pure ALOHA
b) Slotted ALOHA
c) Carrier Sense Multiple Access (CSMA)

2.3.1 Pure ALOHA:

Pure ALOHA was first used in ALOHA system. In this scheme a user transmits a packet as soon as it is ready, hoping that it will not collide with any other packet transmission. If within some appropriate time out period, the transmitting station receives an acknowledgement from the destination then he knows that no conflict occured otherwise he assumes a collission occured and must retransmit later at same randomly chosen time.

Suppose we are given a data traffic source with

T = average interarrival time between messages

δ = average message delay

C = channel transmission rate in bits per second

P = average no. of bits in transmitted data block

N = number of users

Then channel throughput is defined to be the ratio of the rate of successfully transmitted data blocks multiplied by P to the rate C. The bursty factor $\beta = \delta/T$.

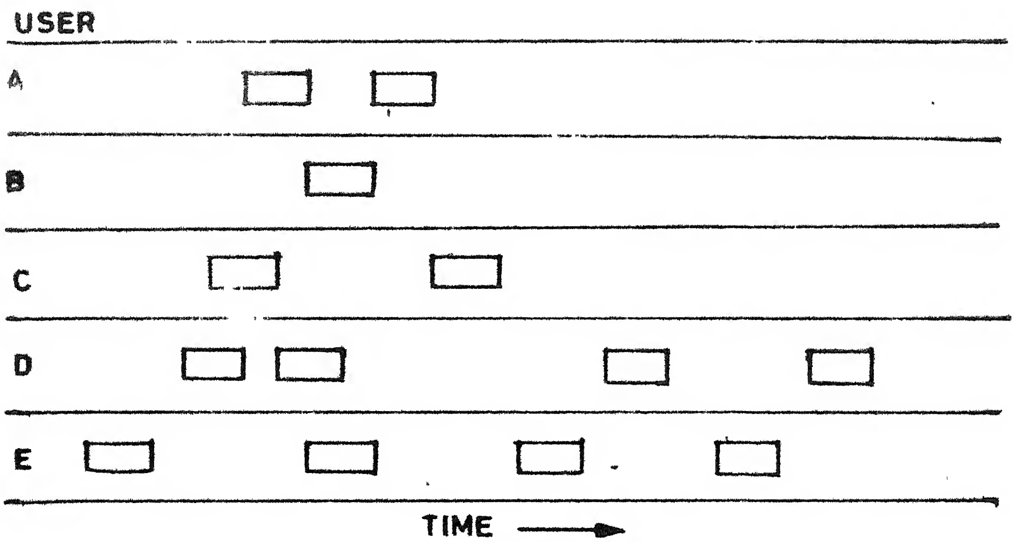
Channel throughput S satisfies $S \leq \beta$. All messages are assumed to consist of single packets. In the limit of an infinite user population $CN \uparrow \infty$ and for each user $\beta \downarrow 0$, the packet birth process is a Poisson process. Abramson [6] made the assumption that the sum of new transmissions and retransmission in the channel (called channel traffic) can be approximated by a Poisson process, which gives rise to the following relationship

$$S = G \cdot e^{-2G}$$

where S = aggregate channel throughput in packets/packet time

G = aggregate channel traffic in packets/packet time

From the above equation, the maximum possible ALOHA channel throughput is obtained at G = 0.5. Thus the ALCOHA channel capacity for an infinite population model is CA=1/e = 0.184.



IN PURE ALOHA PACKETS ARE TRANSMITTED AT ARBITRARY TIMES

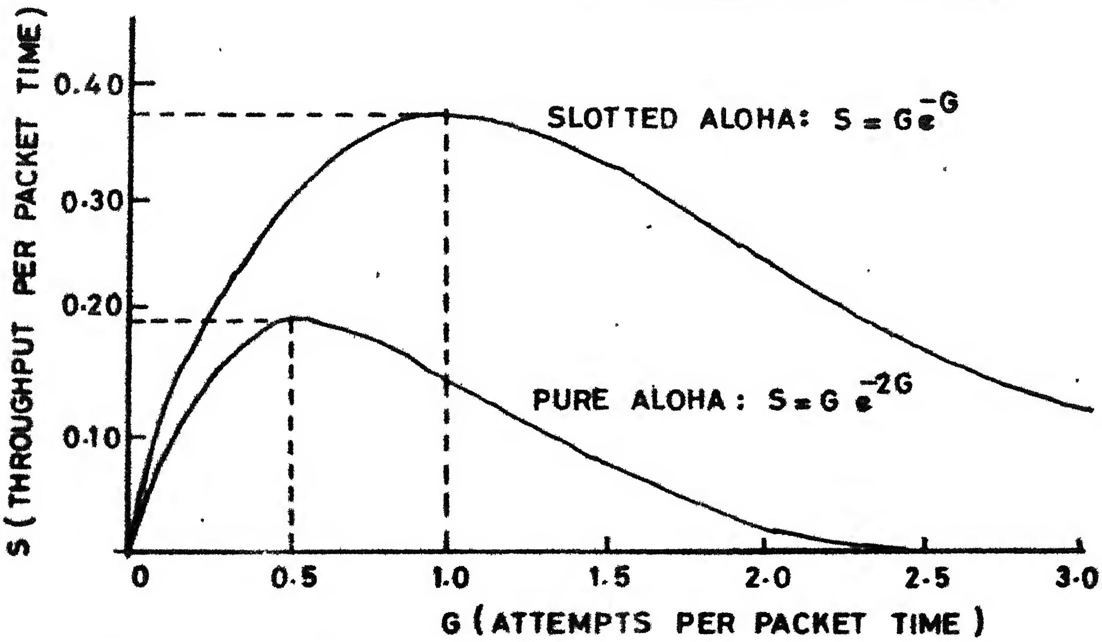


FIG.2.2 THROUGHPUT VERSUS OFFERED TRAFFIC FOR ALOHA SYSTEM.

2.3.2 Slotted ALOHA:

The slotted ALOHA [6] scheme was first proposed and studied by Roberts. In this case, channel users are required to synchronize their packet transmissions into fixed length channel time slots. The protocols of slotted ALOHA are just like ALOHA. However due to slotting, packet collisions due to partial overlaps are avoided. Under same assumptions as above channel throughput and channel traffic is given by

$$S = G.e^{-G} \quad \text{where S is maximized at } G=1$$

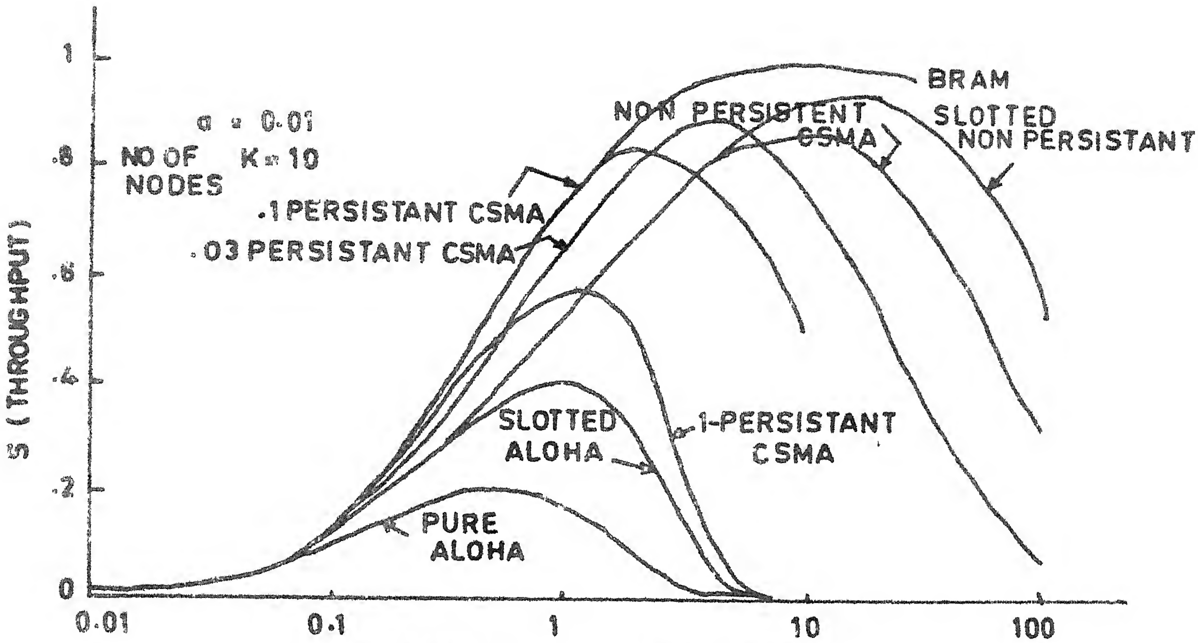
The resulting slotted ALOHA channel capacity for an infinite population model is twice that of the unslotted case

$$C_{SA} = \frac{2}{e} = 0.368$$

Delay throughput characteristics of ALOHA and slotted ALONA are shown in Fig. 2.2.

2.3.3 Carrier Sense Multiple Access (CSMA):

In the packet radio environment the propagation delay between any source destination pair is very small compared to packet transmission time. In such an environment one may attempt to avoid collisions by listening to the carrier due to another user's transmission before transmitting



THROUGHPUT FOR VARIOUS RANDOM ACCESS PROTOCOLS

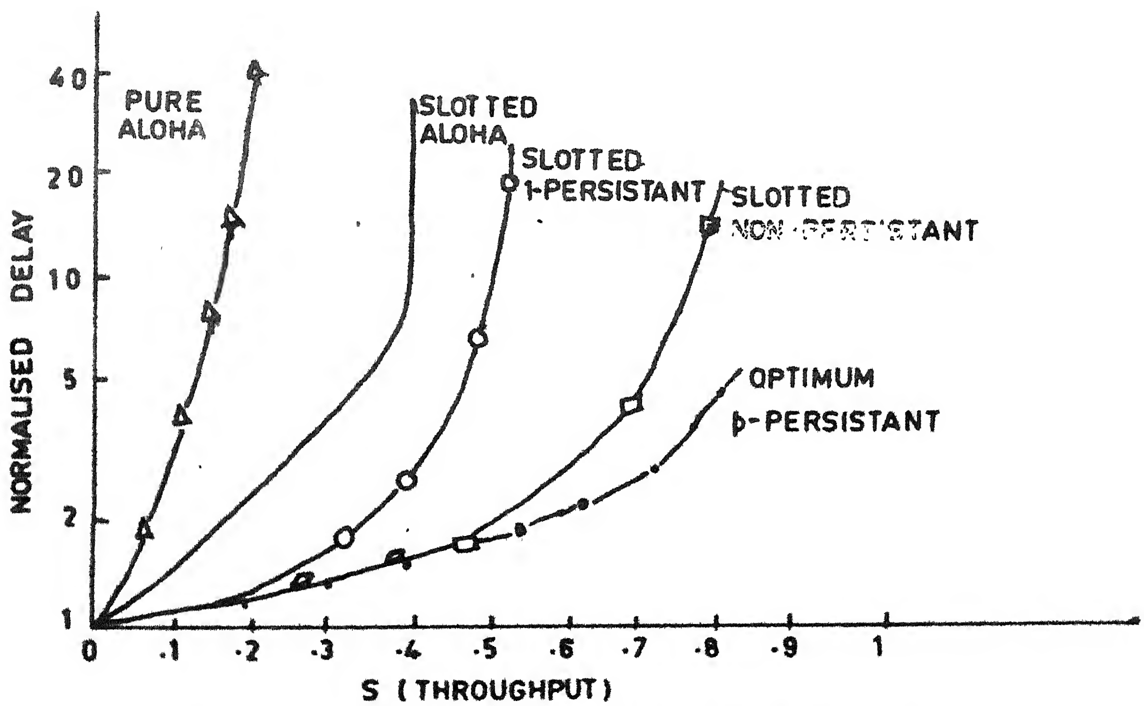


FIG.2.3 CSMA AND ALOHA : THROUGHPUT- DELAY TRADE OFFS (From simulation)

and inhibiting transmission if channel is sensed busy. There are two main CSMA protocols [9] known as non-persistant CSMA and p persistant CSMA. In nonpersistant CSMA, a ready terminal senses the channel and operates as follows:

1) If the channel is sensed idle, it transmit the packet.
2) If the channel is sensed busy, then the terminal schedules the retransmission of the packet to some later time according to the retransmission delay distribution.

The p persistant protocol consist of the following: The time axis is minislotted and the system is synchronized such that all terminals begin their transmission at the begining of a minislot. If a ready channel senses the channel idle, then with probability p, the terminal transmits the packet; and with probability 1-p, the terminal delays the transmission of the packet by τ seconds. If at this new point in time, the channel is still detected idle, the same process is repeated. If the ready terminal senses the channel busy, it waits until it becomes idle and then operates as above. Fig. 2.3 shows the throughput for various random access schemes.

2.4 CENTRALLY CONTROLLED DEMAND ASSIGNMENT TECHNIQUES:

We have so far discussed the two extremes in the bandwidth allocation spectrum as far as control over the user's access right is concerned, the tight fixed assignment,which has the most rigid control; and random access which involves no control, is simple to implement, is adaptive to varying demands but in some situations can be wasteful of capacity due to collisions. We now examine demand assignment techniques which require that explicit information regarding the need for the communication resource be exchanged. We have three such protocols [8] called Polling systems, Adaptive polling or Probing systems and Split Channel Reservation Multiple Access.

2.4.1 Polling:

In this a central controller sends polling messages to the terminals, one by one, asking the polled terminals to transmit. For this the station may have polling list giving the order in which terminals are polled. If the polled terminal has some thing to transmit, it goes ahead, if not, a negative reply (or absence of reply) is received by the controller, which then polls the next terminal. Polling is efficient only if

1) the round trip propagation delay is small

2) the overhead due to polling message is low

and 3) the user population is not a large bursty one.

In this scheme the channel utilization can reach 100 percent of the channel bandwidth if the terminals are allowed to employ their buffers when they are polled. But as a result the variance of packet delay can become in tolerably large.

2.4.2 Adaptive Polling or Probing:

Primary limitation of polling in lightly loaded system is the high overhead incurred in determining which of the terminals have messages. In order to decrease these overheads probing has been proposed [15]. In this the controller interrogates all terminals asking if any of them has a message to transmit, and repeats this question until some terminals respond by putting a signal on the line. When a positive response is received, the central station breaks down the users into subsets (according to some tree structure) and repeats the question to each of the subsets. The process is continued until the terminals having messages are identified.

2.4.3 Split channel Reservation Multiple Access:

In this the central scheduler manages a queue

of requests and inform the terminal of its allocated time. Since the channel is the only means of communication among terminals, and the contention of request packet and data packets is same, in order to prevent collisions,the available bandwidth is either time divided or frequency divided between the two types of data. In split channel reservation multiple access (SRMA), frequency division of a ground radio channel is considered [15] with this configuration there are many operational modes e.g. request/answer to request/message scheme, request message scheme (RM).

RAM: In this the bandwidth allocated for control is further subdivided into the request channel and answer to request channel. The request channel is operated in a random access method (ALOHA or CSMA). Upon correct reception of request packet, the scheduling station computes the time at which the backlog on message channel will be empty and transmits an answer packet back to the terminal, on answer to request channel, containing the address of the terminal and the time at which it can start transmission.

In the RM scheme we have request channel and message channel. When correctly received by scheduling station, the request packet joins the request queue. When the message channel is available, an answer packet is

transmitted (containing ID of queued terminal) by the station on the message channel. After hearing its own ID repeated by the station, the terminal starts transmitting its messages on the message channel.

2.5 PRIORITY BASED PROTOCOL:

In these protocols the various users are attached various priorities. We have four such protocols, Head of line (HOL), Alternating Priorities (AP), Round Robin (RR) and Random Order (RO) [16].

In this scheme all packets are assumed to be of constant length and are transmitted over an assumed noiceless channel. All users have the ability to sense the carrier and the probability of false carrier detection is considered negligible. Let there be N users and τ be the maximum propagation time between any source destination pair. A slot consist of three parts

1) an overhead of (N-1) minislots each of duration τ
2) packet transmission time of duration P
3) one minislot (of duration τ) which accounts for the time between the end of transmission and end of reception.

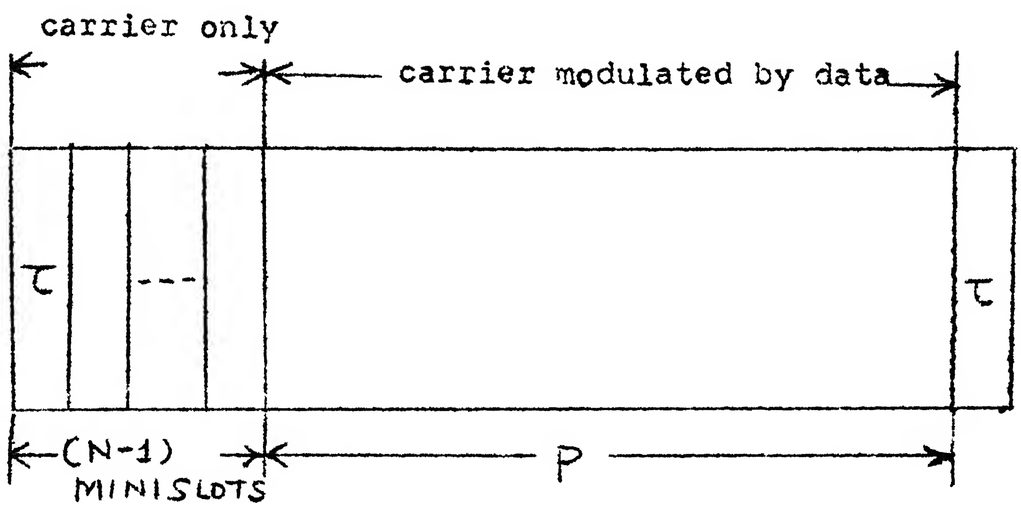


Fig. 2.4: Minislotted Packet

N users are ordered in each slot (of duration $P + n\tau$) by the priority rule which characterizes the protocol. For all priority rules, N users are synchronised in each slot as follows:

1) If the highest priority user is ready, he need not sense the channel and synchronizes his packet's transmission as follows:

   a) At the beginning of the slot, he begins transmission of the carrier. After one minislot at most, all other users know whether the slot is reserved or not.

   b) (N-1) minislots later he transmits his data packets, otherwise if he is idle he remains quiet until the end of the slot.

2) If the ith user in priority ($1 \leq i \leq N$) is ready, he senses the channel for (i-1) minislots.

   a) If no carrier is detected after (i-1) minislots then at the beginning of ith minislot he transmits his carrier and N-i minislots later he transmits his packet.

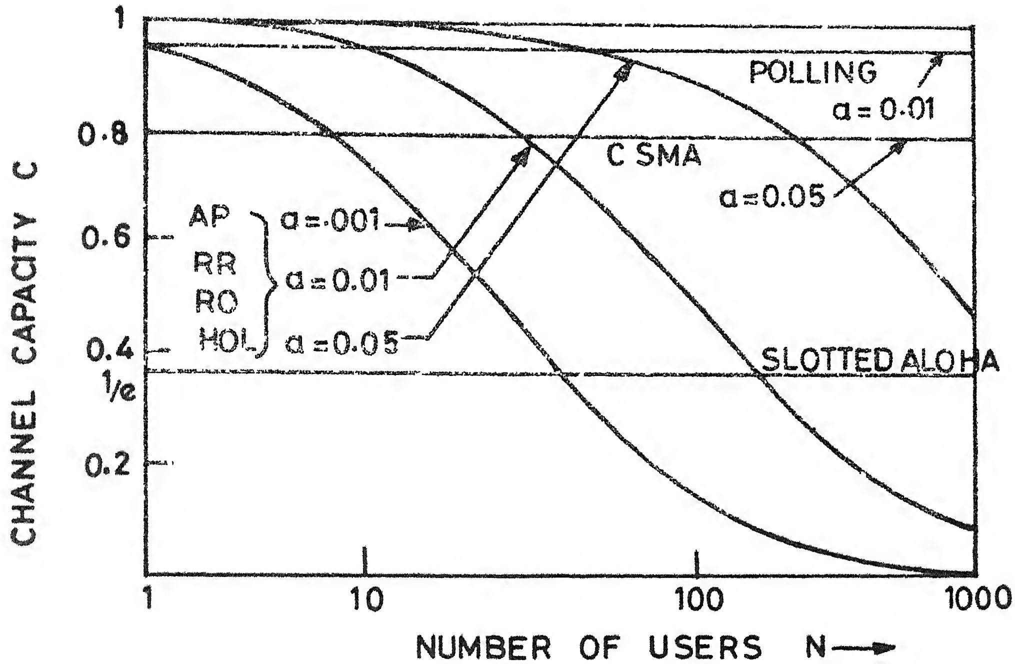
   b) Otherwise he waits for the next slot and process is repeated.

2.5.1 Head of line (HOL) Protocol:

This protocol is devised for a population of N users, on which a fixed priority structure is imposed. The priority among users remains constant in time i.e. ordering of N users does not change from one packet transmission to the next packet.

2.5.2 Alternating Priorities Protocol (AP):

In this protocol once a user seizes the channel, he keeps transmitting packets until he goes idle. Precisely AP obeys following rule. The N users are numbered in a given sequence (say 1,2,3,...N). The highest priority is assigned to that user who transmitted the last packet. Priority then decreases in cyclic order around the numbered users i.e. i, [i mod N]+1, [(i+1) mod N]+1,... [(i+N-2) mod N] + 1.



EFFECT OF NUMBER OF USERS ON CHANNEL CAPACITY

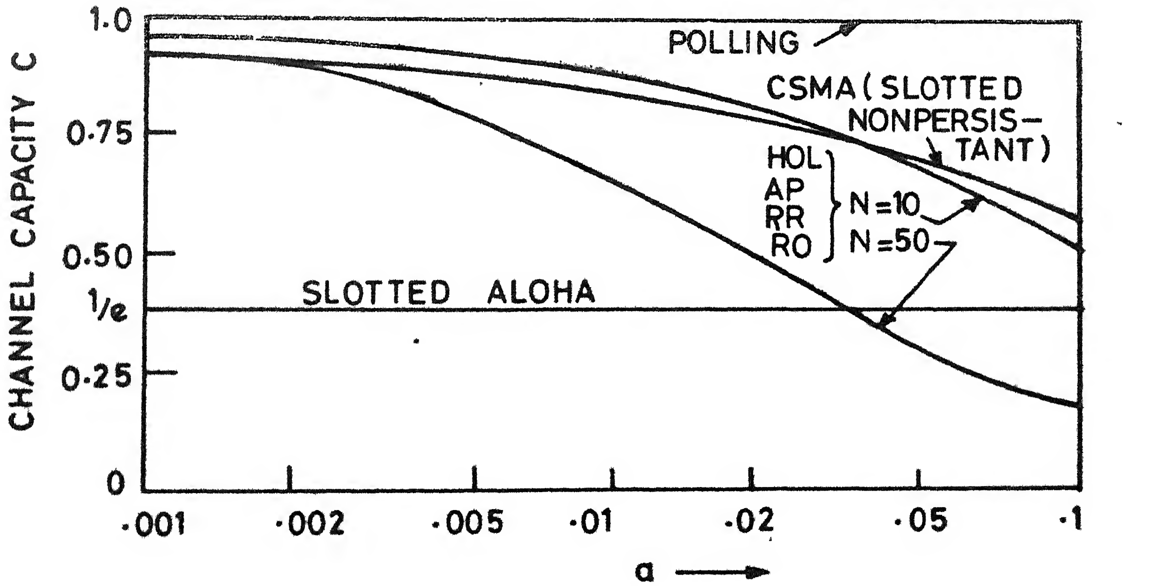


FIG.2.8 EFFECT OF PROPAGATION DELAY ON CHANNEL CAPACITY

2.5.3 Round Robin Protocol (RR):

The users are numbered according to a given sequence (say 1,2,...N). The highest priority is assigned in a round robin cyclic fashion among users. That is the highest priority is assigned to that user whose number (mod N) follows that of the user who had highest priority in the previous slot. This is true even if user i was idle in the previous slot.

2.5.4 Random Order Protocol (RO):

In this each user generates the same pseudo random permutation of 1,2,...N which gives the priority of N users for the current slot. No matter who uses the current slot, each user generates a new permutation which gives priority order of N users for next slot. Throughput delay characteristics of these protocols are given in Fig. 2.5.

2.6 DEMAND ASSIGNMENT MULTIPLE ACCESS PROTOCOLS:

More recently, a number of new demand assignment multiple access (DAMA) protocols [17] have been proposed for broadcast networks. These schemes provide conflict free transmission using distributed access protocols with round robin scheduling functions which thus lead to bounded delay. There are three possible access mechanisms according to which these are classified. These are

a) scheduling delay access

b) the reservation access

c) the attempt and defer access

for packet radio system only scheduling delay access protocols are suitable. In this each station is assigned a unique index number. These indices form a logical ring which determines the order in which stations are allowed to transmit. Included with each transmission is a field for the index number of sending station. Let S_i be the station currently transmitting and EOC(i) denotes its end of carrier. This scheme is implemented in following steps.

a) Let us assume station S_j wants to transmit. Station S_j detects the carrier.

b) On detecting EOC(i), it assigns itself a scheduling delay $H_j(i)$, function of both i and j, according to which it schedules its transmission after time $H_j(i)$.

c) $H_j(i)$ is so selected such that, if at least one of the stations with indices between S_i and S_j wants to transmit, then that station which is next in sequence following S_i would have begun to transmit its packet.

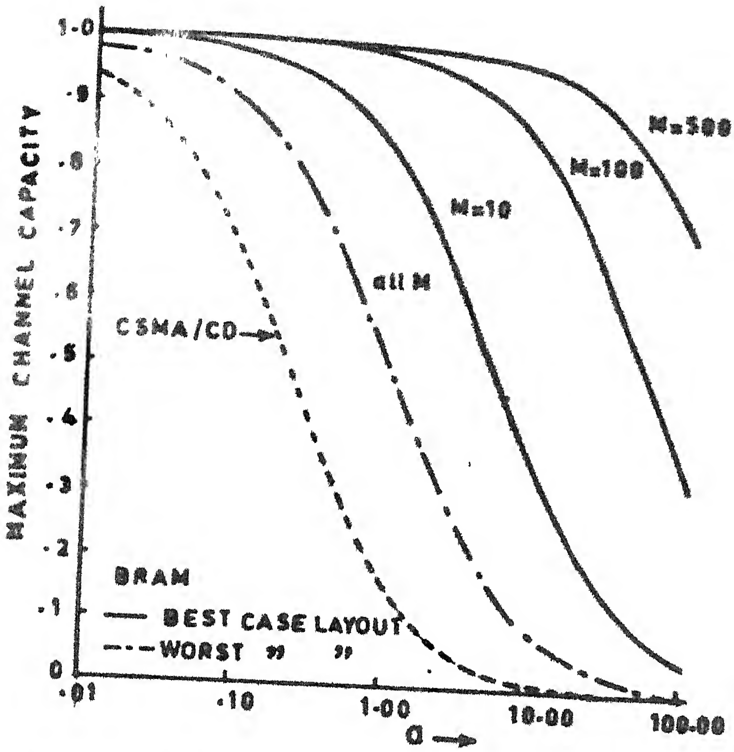


FIG. 2.6(a) NETWORK CAPACITY VS a For BRAM

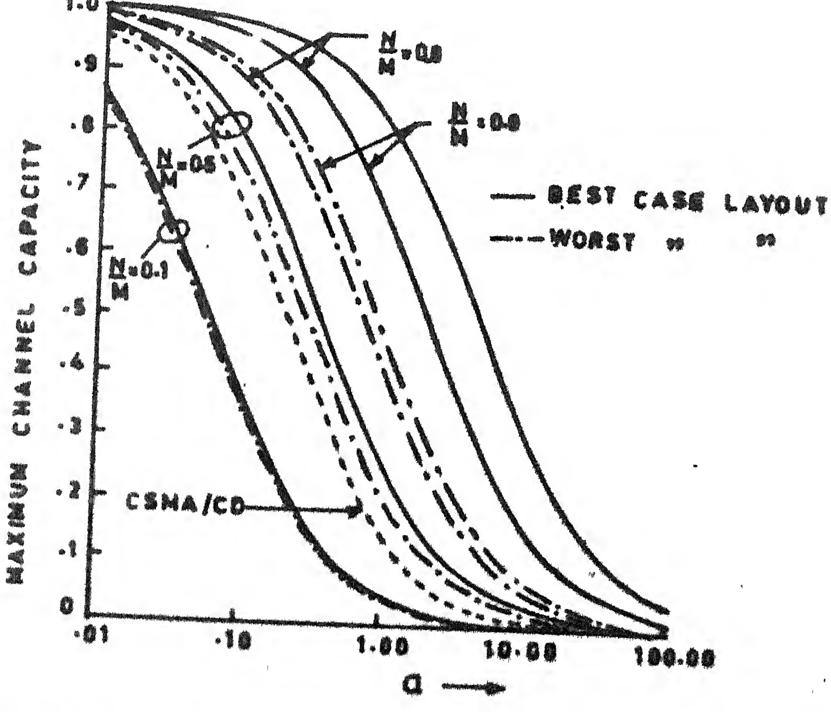


FIG. 2.6(b) NETWORK CAPACITY VS a (when some stations

d) This station will be detected by S_j before scheduled transmission of S_j thus resulting in round robin scheduling.

The various protocols which use this access mechanism are Broadcast Radio Access Mechanism (BRAM), Minislotted alternating priorities (MSAP), Source synchronised access method (SOSAM).

2.6.1 BRAM: This protocol allocates the channel to nodes via a decentralized protocol. In this we assume that there exist independent acknowledgement channel and nodes in network can detect the busy/idle status of channel in a negligible time. BRAM act as follows:

1) The station which wants to transmit, senses the channel, if found idle, then it transmits in next slot.

2) If the channel is sensed busy then it waits for the channel to become idle so that time for giving scheduling delay is known, after which it returns to execute step 1.

Fig. 2.6 gives throughput versus offered traffic rate for BRAM.

2.6.2 MSAP: This is carrier sense version of polling suitable for a small number of data users [15]. In this time axis is slotted with the minislot size equal to maximum propagation

delay. All users are synchronised and may start transmission only at the beginning of a minislot. Users are ordered from 1 to M. When a packet transmission ends, the alternating priorities (AP) rule assigns the channel to the same user who transmitted the last packet. (say user i) if he is still busy; otherwise the channel is assigned to the next user in sequence i.e. (i mod M+1). The latter (and all other users) detects the end of transmission of user i by sensing the absence of carrier over one minislot. At this new point in time either user (i mod M+1) starts transmission of a packet or if he is idle in which case a minislot is lost and control of channel is handed over to the next user in sequence.

2.6.3 Source Synchronized Access Method (SOSAM):

In this scheme, access method is similar to BRAM. To accomplish this, all stations must have explicit knowledge of the propagation delay between every pair of users. Given this knowledge, S_j can determine the minimum time required after detecting EOC(i), to detect a potential transmission from S_{j-1} and set $H_j(i)$ to this amount of time. The throughput in this case is same as for BRAM and is given in Fig. 2.6.

2.7 MULTI-ACCESSING TREE PROTOCOL:

The tree algorithm as given by Capetanakis [18] is stated below. First, some definitions concerning tree graph are given.

Depth of a node - the number of branches between the node and the root node. The root node is at depth zero.

Degree of node - The number of branches that emanate from a node.

Subtree T_{ij} - the binary rooted subtree whose root node is n_{ij}, where, in a binary rooted tree, j corresponds to the particular one of the 2^i nodes at depth i.

Let each user correspond to a leaf on a binary tree as shown in Fig. 2.7.

In this each user has a four bit binary address as shown in Fig. 2.7 . Also, let T_x and T_y be two binary rooted subtrees and assume that no collisions have occurred up to the begining of the present pair of slots. Then binary tree algorithm is as follows:

1. Choose $T_x = T_{10}$; $T_y = T_{11}$
2. Transmit all the packets in T_x in the first slot of the present pair slots, and transmit all the backets in T_y in the second slot.

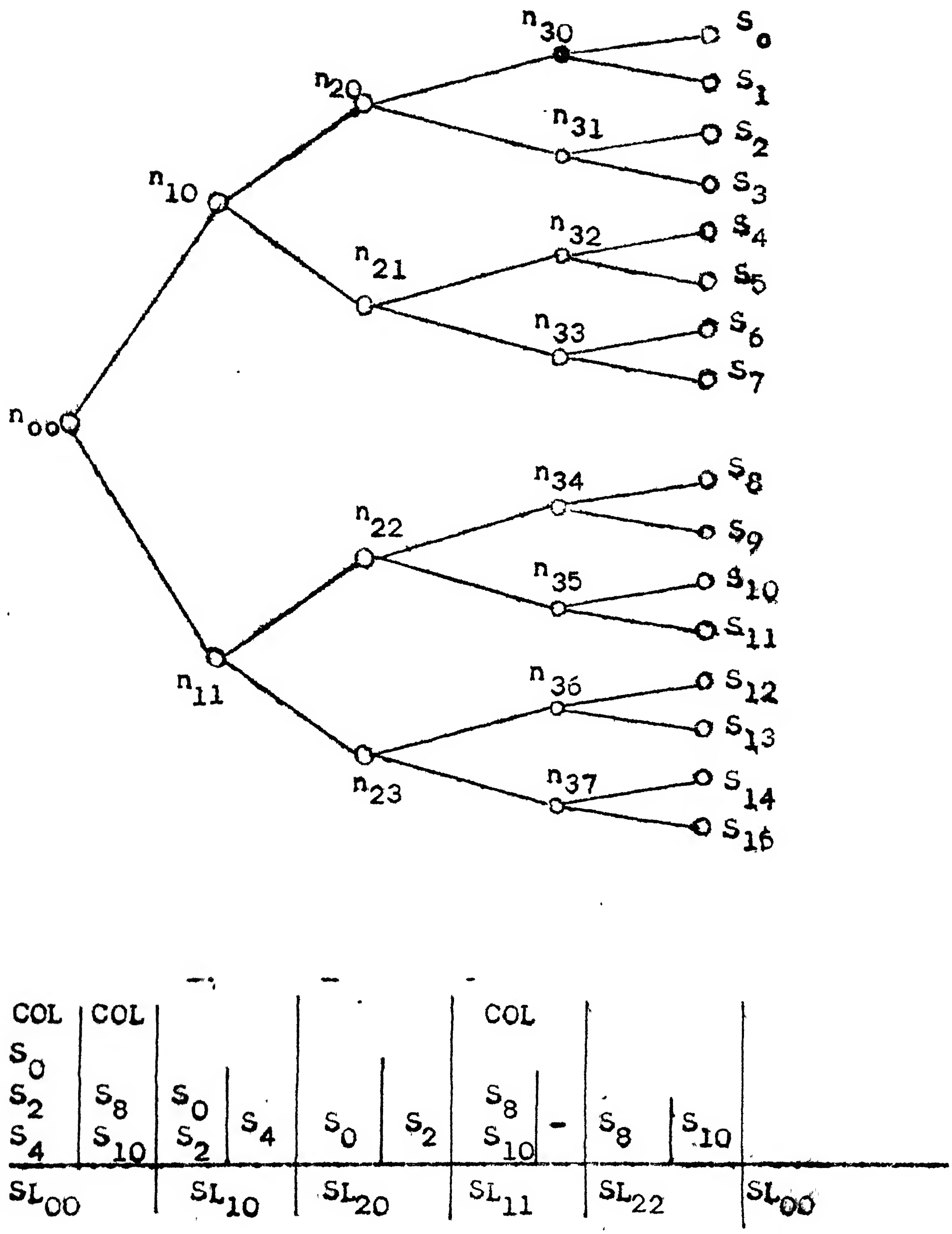


Fig. 2.7 16 Leaf Binary Tree

3. If any collision occur in the preceding step, then

   a) until these collisions are resolved, do not transmit any new packets unless they arrive at sources which either did not have an opportunity to transmit or they did and the transmission resulted in a still unresolved conflict.

   b) Resolve the first collision (if any) before resolving the second (if any).

A collision in T_x (or T_y) is resolved by dividing T_x (or T_y) into two halves (say A and B), setting $T_x = A$, $T_y = B$ and then repeating ~~step 2 and 3~~.

Let us explain this with an example. Let there be a 16 users $S_0, S_1, \ldots, S_{15}$ and let each correspond to a leaf on the 16 leaf binary tree as shown in Fig. 2.7 Slots are paired and a slot pair is designated by SL_{ij}. Assume that no collisions have occured until begining of SL_{oo} when sources S_0, S_2, S_4, S_8 and S_{10} each has a packet to transmit. The begining with SL_{oo}, where the first contention arises, the tree algorithm takes following steps.

1. All the sources which underwent collision generate random no. 10 or 11.

2. All the sources in T_{10} (say S_0, S_2 and S_4) transmit packet in first slot of SL_{oo} and corresponding sources in T_{11} do so in the second slot.

3. This results in two collisions, one among S_0, S_2 and S_4 and other between S_8 and S_{10}.
4. Since there was atleast one collision in SL_{oo} , any new packets that are generated are not transmitted until the old contention is resolved.
5. Since there was a collision in T_{10}, the sources at T_{10} are divided in half and the packets in T_{20} and T_{21} are transmitted in the first and second slot of SL_{10}.
6. This results in a collision between S_0 and S_2 and in a successful transmission by S_4.
7. Since there was a collision in T_{20}, T_{30} and T_{31} transmit their packets in the first and second slot of SL_{20}.
8. This results in successful transmission by S_0 and S_2.
9. Since there was a collision in T_{11}, T_{22} and T_{23} transmit their packets in succession. This results in a collision between S_8 and S_{10} in the first slot and no transmission in the second.
10. Since there was a collision in T_{22}, T_{34} and T_{35} transmit. This results in two successful transmissions by S_8 and S_{10}.

This system has a maximum average throughput of .43 packets/ slot [18,19].

After having a review of all the protocols for a packet radio we have suggested a protocol for packet radio under design in the next chapter.

CHAPTER 3

DESIGN OF PACKET RADIO USING RF VOICE LINK

3.1 CHOICE OF PROTOCOL FOR PACKET RADIO

Basic operation of a packet broadcast network has been explained in Chapter 1. A single broadcast channel is shared among a population of distributed users. It is assumed that each user is capable of sending and receiving data at the channel transmission rate of C bits/sec. Data messages are segmented into fixed length packets for transmission. Each packet contains its own address and destination address (es) as well as parity bits for error detection. A packet transmitted successfully by any user i.e. in the absence of errors due to noise or interference from another user,will arrive correctly at all users. The packet will be accepted by the intended receiver and ignored by others. When packets transmitted by different users collide in the channel it is assumed that none of the packets involved in a collision will arrive correctly at the intended receiver. Such collisions are detected as transmission errors.

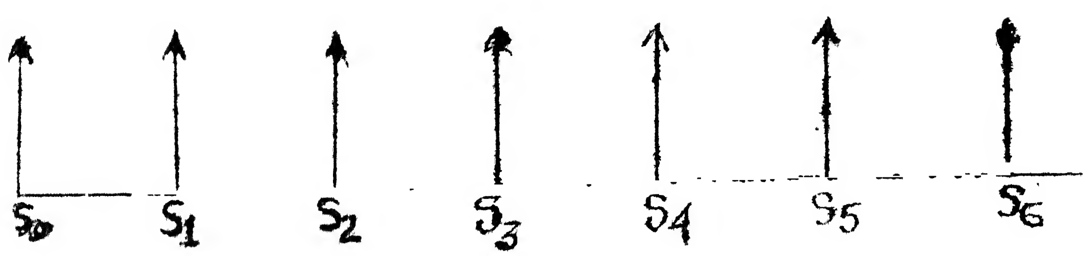
A protocol based on multiaccess tree protocol, as described in last chapter is suggested below for the packet

radio. Let a large number of users generate messages in a Poisson manner at a total rate of λ messages per unit of time starting at time 0. In order to control, the start time of a packet, the master sends a RF burst and the station which wants to transmit detects this RF burst and starts his transmission after giving a delay of few miliseconds. This is required for clock synchronisation. We explain this protocol in following steps.

1. If only one user transmits, the transmission is successful.

2. If transmission from two or more users overlap a collision is said to occur and all messages are lost and must be retransmitted at a later time.

3. When the users collide say at S_0, refer Fig. below, they generate a binary 1 or 0. All those users under contention which generate a 1, transmit in the next slot i.e. S_1 and all those who generate a 0, transmit in the next to next slot i.e. S_2.



Case 1 If there is no collision in S_1:

4. It means users in S_2 will undergo collision and they must again generate a binary number 1 or 0.

5. All those who generate 1, will transmit in next slot of time i.e. S_3. All those who generate a 0 will transmit in S_4.

6. If there is only one user in slot S_3, he will be successful in his transmission, otherwise they again generate a binary number and transmit in S_5 or S_6 and this goes on till all users in S_3 are successful.

7. After users in slot S_3 are successful, users in slot S_4 generate binary numbers 1 or 0 and transmit in next slots till their contention is resolved and are successful.

Case 2 If there is collision in S_1:

8. Since there is a collision in slot S_1, the users in S_1 again generate binary 1 or 0 and all those who generate 1 transmit in next slot.

9. All users who generate 1 will observe the channel. If channel is sensed idle i.e. no user generated a 0, then all users in S_1 who generate 1 will transmit in S_2, otherwise they transmit in slot S_3.

10. This process goes on as explained above till all the users are successful. After resolving this contention, if any new users is ready with the packet he goes ahead and the process is repeated.

In this protocol throughput behaves roughly like 0.487-p, where p is the probability of false collision indication [19].

3.2 CHOICE OF A RADIO CHANNEL

A VHF set which will operate in the lower VHF region of frequency spectrum has been selected as it is expected, that getting a license will be possible for a small channel out of this band, all other bands being very much overcrowded. We select a VHF, FM radio transmitter/receiver which is in use by army. The various characteristics of this radio set are given below.

3.3 FM RADIO CHARACTERISTICS

It is assumed that a standard VHF, FM radio set ANPRC 25 being used in army (or equivalent) is available to us. The characteristics of this radio set are

Frequency Range:

Low Band 30.00 to 52.95 MHz

High Band 53.00 to 75.95 MHz

No. of channels	920
Channel spacing	50 KHz
I.F.	11.5 MHz
Type of modulation	FM
Type of transmission and reception	Voice
Transmitter output power	1.1 to 1.6 W
Range	8 km
Power source	Battery dry HT/LT supplying 15/3v 1 Amp.
Power consumption	Receive 0.7 W
	Transmit 16.5 W
Frequency deviation	10 Kc/sec

Military FM radios do not operate in an isolated environment. Instead they are part of networks in which each radio has a specific frequency allocation. For purpose of standardisation VHF FM band is divided into 50 KHz wide channels. This can be used for both analog speech and 16K bit/sec data. They are designed to pass the spectrum of analog voice.

Radio is divided into two functional parts namely the transmitter and the receiver. A block diagram of transmitter /receiver is shown in Fig. 3.1. The transmitter

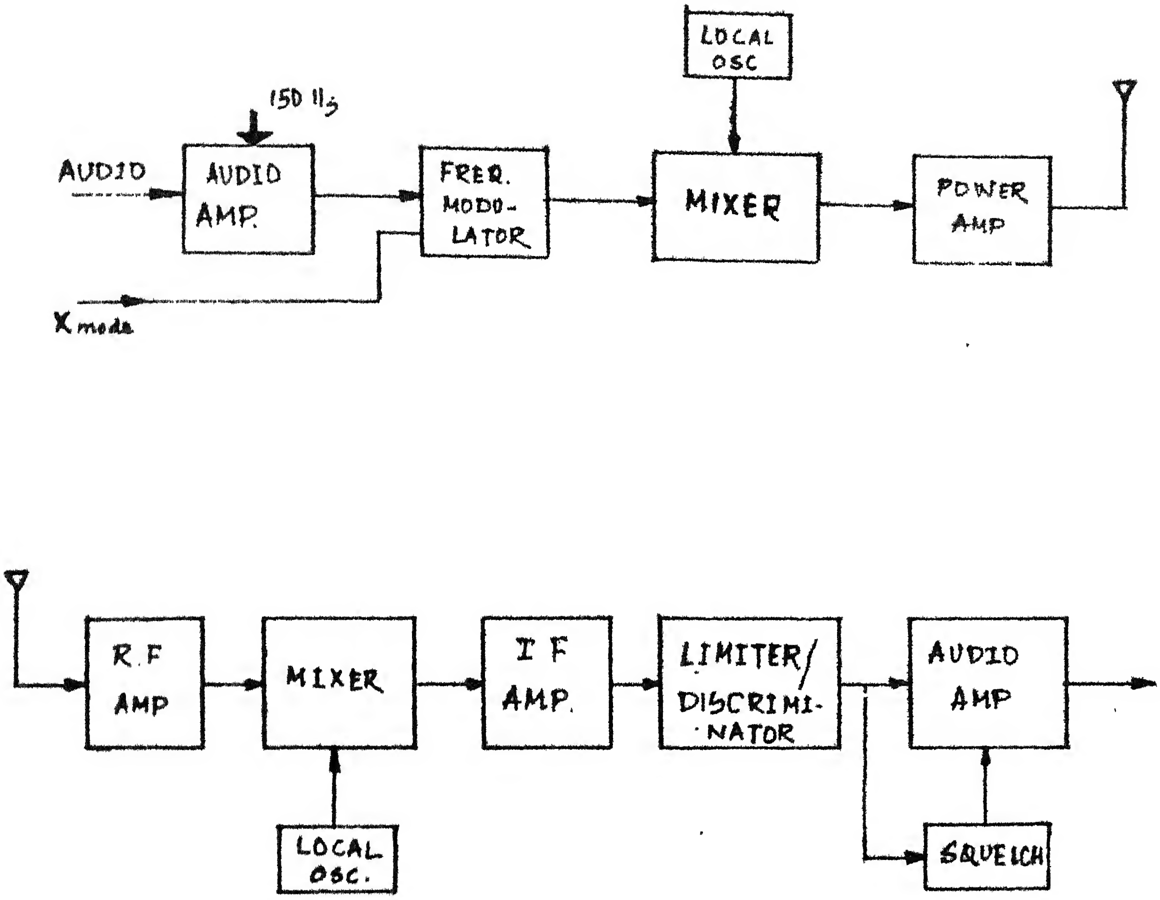


Fig 3.1 FM RADIO TRANSMITTER/RECEIVER

has two operating modes, the narrow band or normal audio mode and wide band or X mode. Both modes are energised by the operator pushing the push to talk switch thereby turning on transmitter power, disconnecting receiver from antenna and connecting transmitter to antenna. In the normal audio mode the input goes through the audio amplifier where it is filtered to 500-3500 Hz band. The amplifier sums the filtered audio signal with a 150 Hz side tone used at receiver for squelch . The amplifier output then derives frequency modulator whose output is hetrodyned to RF by mixing with local oscillatorat 11.5 MHz and then amplified and transmitted. In X mode the output signal by passes the audio amplifier and drives the frequency modulator directly. The bandwidth of X mode input is normally 30 Hz to 16 KHz and 150 Hz side tone is not generated.

The receiver performs inverse of transmitter operations. The received signal is amplified at RF and mixed to IF where it is further amplified and band pass filtered. The IF amplifier output drives a limiter discriminator with an output bandwidth extending from 20 Hz to 16 KHz. The discriminator is thus able to demodulate both the normal audio and X mode.

Future needs will require that these radios be used to transmit base band digital data at both low rates 75 bits/sec to 4800 bits/sec and high rates 2400 bits/sec to 16,000 bits/sec. At the present time, there is essentially no data available to the performance of these radios when transmitting digital data. In order to get such data, a means of converting baseband data into a format, which can be utilized by these radios is required. Modified doubinary is a step in this direction which helps in reshaping the baseband digital signal.

3.4 MODIFIED DUOBINARY SCHEME

Modified duobinary or partial response signalling [20,21] is a type of filtering that meets the objective of efficient digital transmission. If a rectangular low pass filter is used for band limiting baseband digital data, Nyquist rate of 2 bits/sec/Hz can be achieved, but it is very difficult to realise such a filter. Nyquist showed that for a data rate of 1/T bits/sec, a bandwidth of $\frac{1}{2T}$ Hz is sufficient.

Consider that data pulse train at 1/T pulses/sec is being transmitted on the channel. The channel shapes these pulses, providing pulses of considerable tails, which interfere producing intersymbol interference [ISI]. Modified duobinary introduced by Lender [20] uses control amount of

ISI over a span of one, two or more digits causing spectral reshaping. As a result, for a given bandwidth and power input to the channel, it is possible to achieve Nyquist rate. Due to the correlation between digits, the pulse train has distrinct pattern which can monitor error conditions of the channel. The transfer function of such a correlative system (transmit and receive) has the following form

$$F(\omega) = \sum_{k=o}^{n} a_k \; e^{-jkwT}$$

where $a_o = 1$, and $a_k = \pm 1$ or 0

3.4.1 Generalised Partial Response System:

Let us consider a pulse train which is passed through a channel with transfer function $H(\omega)$. Let N be the smallest number of contineous samples that span all nonzero samples. Then if f_n, $n = 0,1,2,\ldots N-1$ are these N sample values, the PRS polynomial is

$$F(D) = \sum_{n=o}^{N-1} f_n \; D^n$$

where D is the delay operator.

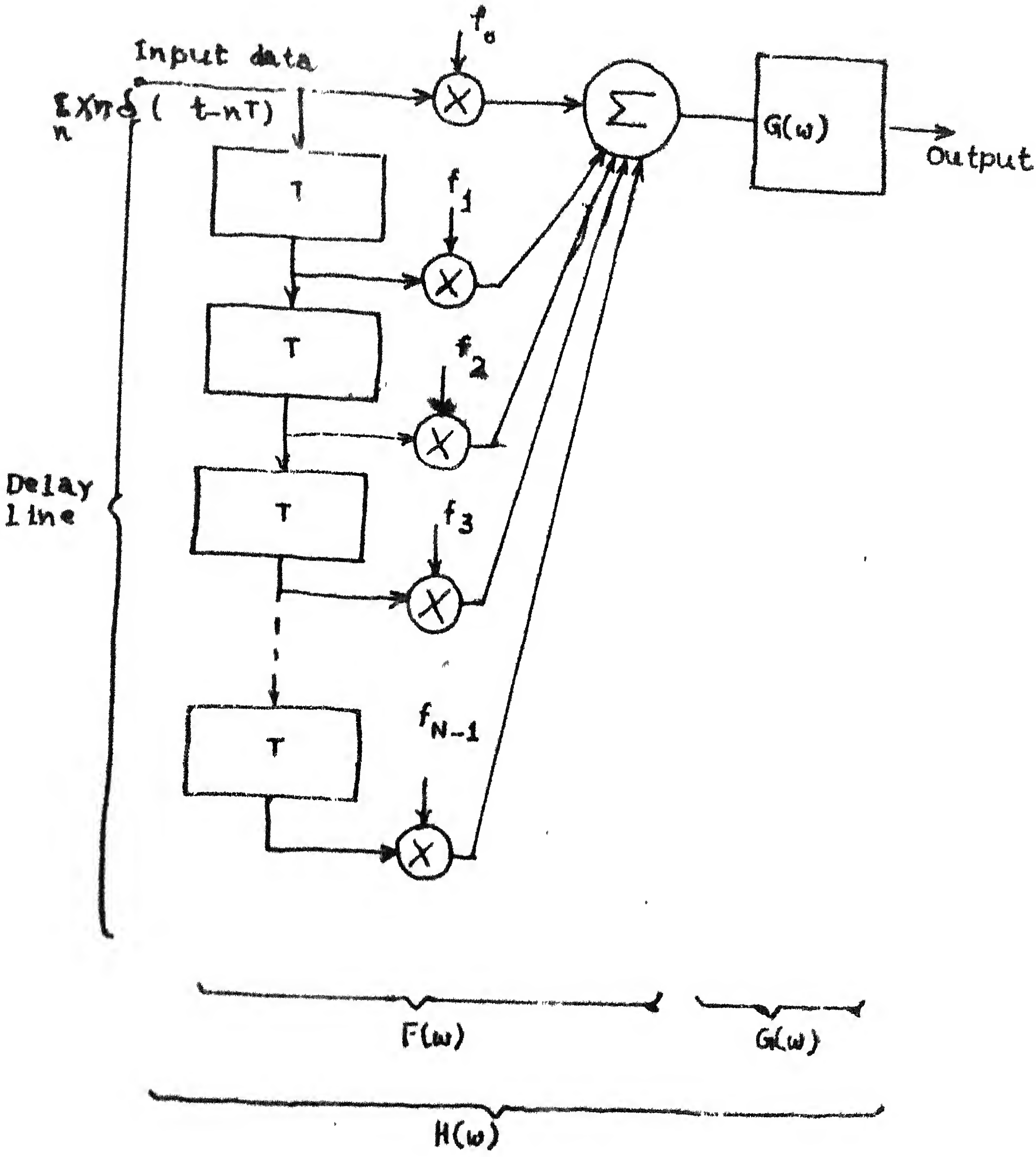


Fig 3.2: GENERALIZED PARTIAL RESPONSE SIGNALLING

For input sequence x_n, the output sequence y_n is given by

$$Y(D) = X(D)\,F(D)$$

$$\text{where}\quad Y(D) = \sum_{n=0}^{\infty} Y_n D^n$$

Fig. 3.2 shows a method of generating the PRS system function $H(\omega)$. It consists of a delay line with coefficients 'f_n'. The added signal passes through a filter with frequency response $G(\omega)$.

$$\text{Then } F(\omega) = F(D)|D = \exp^{(-j\omega T)}$$

$$= \sum_{n=0}^{N-1} f_n \exp^{(-j\omega nT)}$$

where T is symbol spacing.

Based on the generalized PRS system a number of PRS polynomials can be chosen by choice of f_n and $G(\omega)$.

3.4.2 Choice of PRS Polynomial for MDB:

Modified duobinary is characterized by polynomial $(1-D^2)$, where D^2 is delay of 2 bit durations. Kabal and Pasupathy [23] have given a detailed reasoning in choosing the PRS polynomials. The block diagram of MDB is shown in Fig. 3.3. Consider a binary input train consisting of

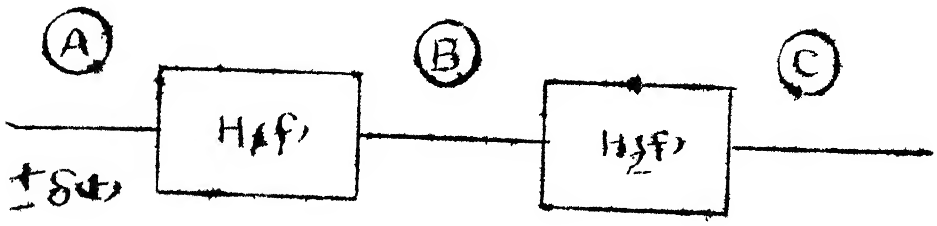


Fig. 3.3: Block diagram of MDB system

1's and 0's, represented by impulses $+\delta(t)$ or $-\delta(t)$ as shown at A in Fig. 3.3 For each $\delta(t)$ input at A output at C is represented by h(t)

$$h_1(t) = \delta(t) - \delta(t - 2T)$$

$h_1(t)$ is impulse response of H (f)

$$h(t) = \frac{\sin \pi t/T}{\pi t/T} - \frac{\sin \pi(t-2T)/T}{\pi(t-2T)/T}$$

$$h_1(t) = \mathcal{F}^{-1}[H_1(f)]$$

$$H_1(f) = 1 - e^{-j4\pi fT}$$

$$= (1 - e^{-j2\pi fT})(1 + e^{-j2\pi fT}) \qquad (3.1)$$

$$= 2 \sin 2\pi fT.$$

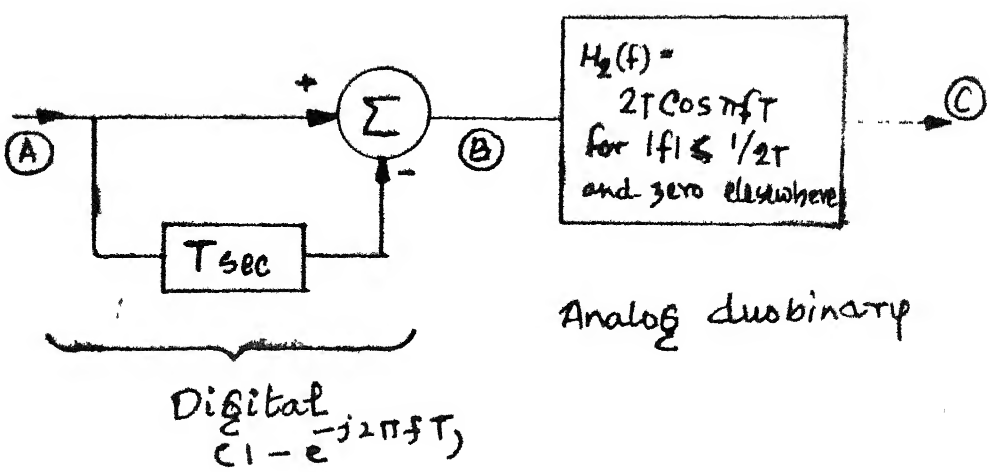


FIG 3.4 MDB IMPLEMENTATION

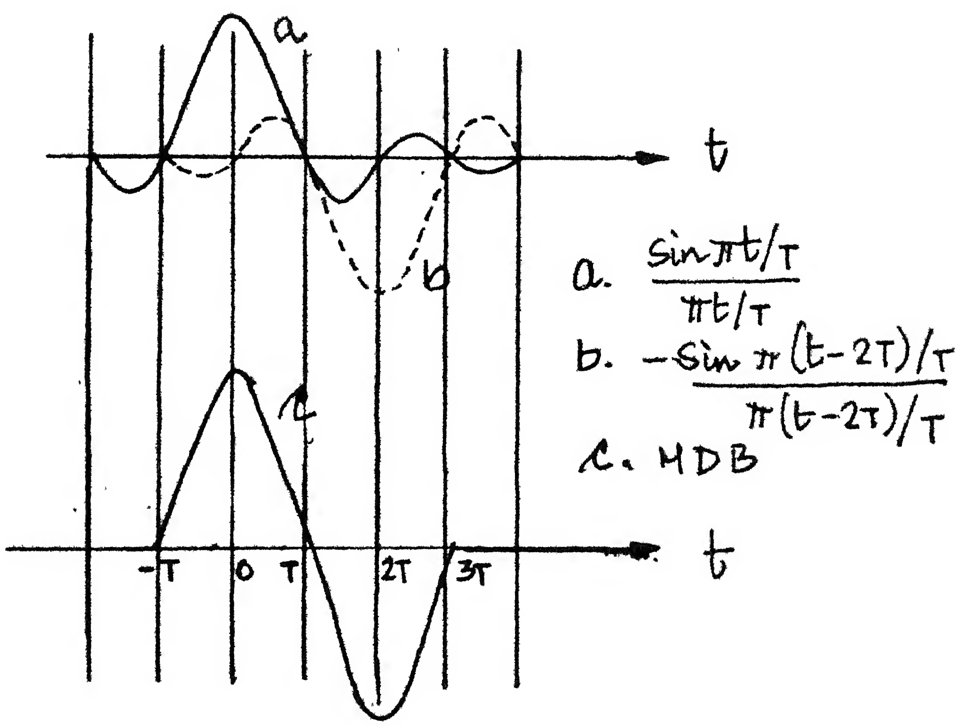


FIG 3.5 IMPULSE RESPONSE OF MDB

and $H_2(f) = \begin{matrix} T & \text{for } f \leq 1/2T \text{ and} \\ 0 & \text{elsewhere} \end{matrix}$

The overall transfer function is given as

$$|H(f)| = |H_1(f) \cdot H_2(f)|$$

$$= \begin{matrix} 2T \sin 2\pi fT & \text{for } f \leq 1/2T \\ 0 & \text{elsewhere} \end{matrix}$$

From equation (3.1) we can see that $1 - e^{-j2\pi fT}$ can be implemented digitally and $(1 + e^{-j2\pi fT})$ can be implemented like a duobinary filter [24]. The implementation is shown in Fig. 3.4 and impulse respons is shown in Fig. 3.5

3.5 MDB ENCODING

The encoding process [20] involves two elementary code A block diagram alongwith wave shapes is shown in Fig. 3.6. Each bit in the waveform at [B] is correlated with the second bit back rather than with the previous bit. The conversion process shown in the base band form is carried out through a filter with bandwidth of $2f_1$ c/sec. The zero frequency component is eliminated and the energy is centered at a frequency of f_1 c/s. It is important to note that signalling rate is $4f_1$b/s as compared to $2f_1$b/s for binary system having the same bandwidth. The resulting

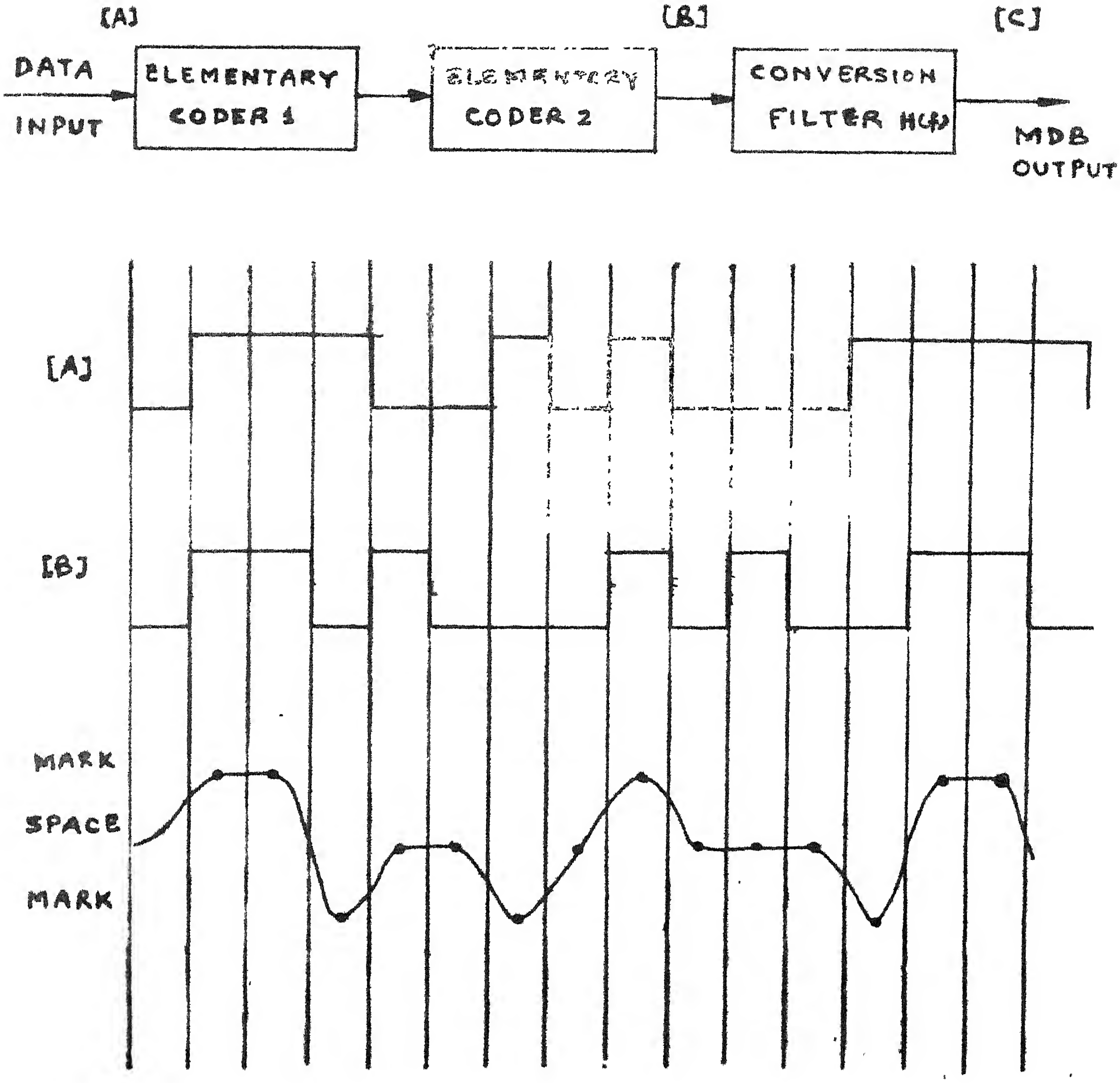


Fig 3·6 MDB ENCODING

waveform at [C] has several interesting characteristics. There is one to one correspondence of each sampling point (indicated by a dot) with the original data at [A] in addition MARKS are always at the extreme levels and SPACES at the centre level, so each digit can be identified independently at the receiving end. As would be expected, owing to its correlation properties, the modified duobinary signal follows a predetermined set of rules. These rules can be easily deduced by grouping all the successive MARKS in pairs and assigning the pair number to each MARK as shown in Fig. A MARK bearing number 1 in a pair of two successive MARKS always has the opposite polarity relative to the previous MARK, which of course carries number 2. The polarity of the MARK that has number 2 relative to the previous MARK bearing number 1 is governed by the set of odd and even rules i.e. if the number of intervening SPACES between a pair of MARKS numbered 1 and 2 is even, their polarities are identical; if not, their polarities are opposite. Such rules permit the detection of errors. Encoding also helps in easy decoding. By sampling at sampling instants we get decoded binary data straightaway.

3.6 PERFORMANCE OF MODIFIED DUOBINARY

By introducing a controlled amount of ISI for wave-

shaping, the partial response signalling technique or MDB allows a practical transmission system to transmit at the Nyquist rate of 2W bits/sec/Hz in a bandwidth of W hertz. Furthermore MDB coding format can detect transmission errors. The performance of modified duobinary system can be evaluated under following heads:

a) Data rates: Duobinary signalling can achieve upto 43% more than the Nyquist rate and MDB can go upto 16% more than the Nyquist rate as given by Kabal and Pasupathy [13].

b) Signal to noise ratio (SNR): Due to the increase in number of levels viz. 3, there is SNR degradation of 3 dB compared to binary. Vertical opening of eye gives SNR degradation. Knowing SNR we can find the probability of error,

$$P_E = f(S/N)$$

c) Spectral shaping: MDB offers good spectral shaping and can be used in band limited channels. MDB has no DC component. This is important since many transmission links can not transmit DC.

A packet radio operates in a bursty environment and hence there is a need that the receiver clock must synchronize with the system clock within a reasonably small time of say 10 milisec. In the next chapter we discuss the hardware implementation of MDB and a scheme for clock synchronisation.

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CHAPTER 4

HARDWARE IMPLEMENTATION OF PACKET RADIO

A block diagram for packet radio transmission and reception is given in Fig. 4.1. The data to be transmitted is available in binary form. First we do the spectral shaping of this binary data into Modified Duobinary form. The MDB waveform (which is in analog form having three levels) is passed through a low pass transmitting filter. This band limited output is fed to transmitter of Radio set which is available and whose specifications are given in Chapter 3. At the receiver end this modulated MDB is demodulated to get MDB. Then from MDB we recover original binary data.

4.1 CONVERTING DATA TO MODIFIED DUOBINARY

Since the system will be operating at low data rate of 4.8 K bits/sec we use analog switches to switch ON/OFF, the signals, based on the logic. The MDB introduces interference after one bit delay and in phase opposition i.e. $1-D^2$. Impulse response of MDB is shown in Fig. 3.5.

$$h(t) = \frac{\sin \pi t/T}{\pi t/T} - \frac{\sin \pi (t-2T)/T}{\pi (t-2T)/T}$$

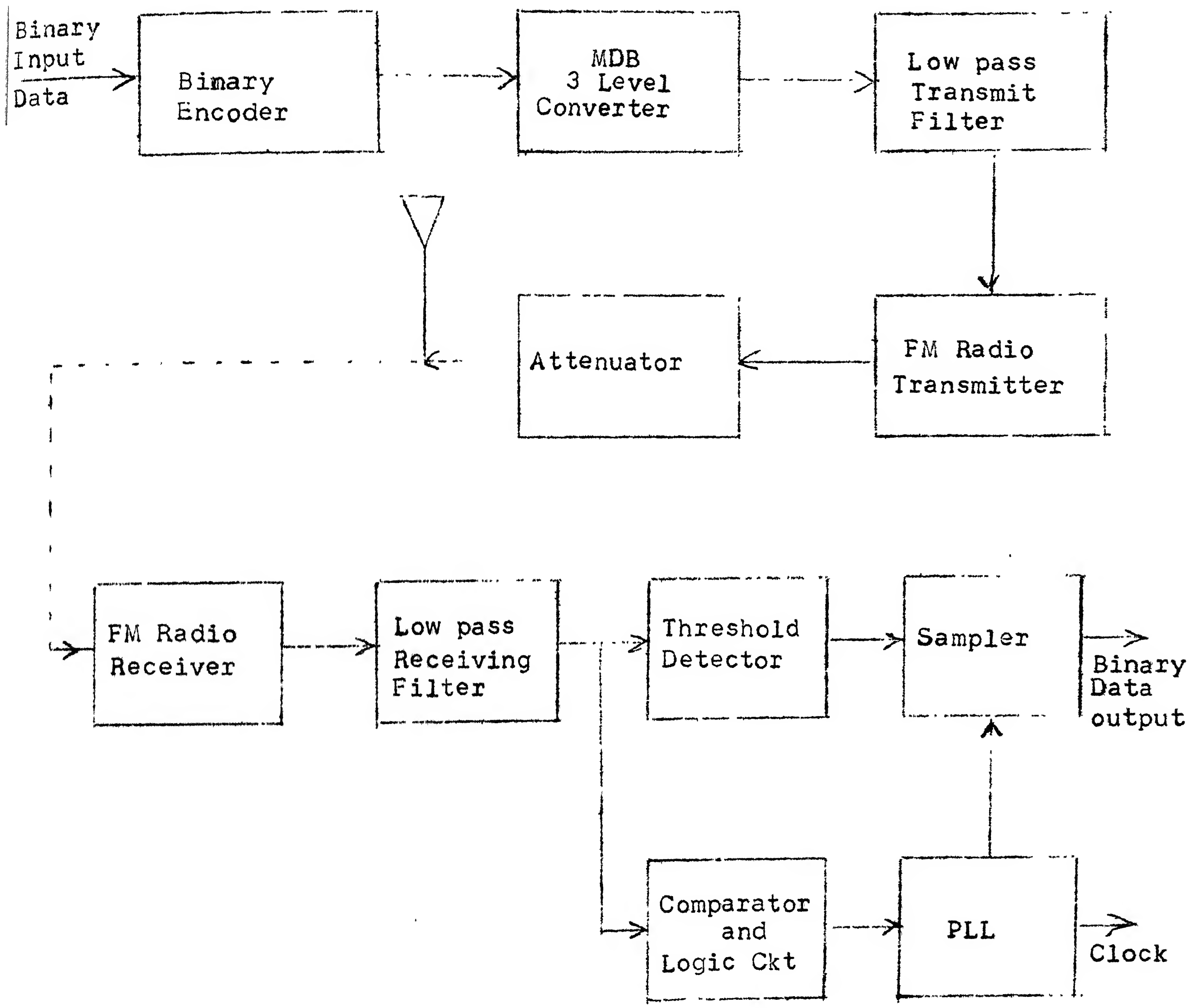


Fig. 4.1: Block Diagram of Packet Radio

The ISI in current bit is due to the bit prior to the past bit. The data streem is split into two streams of odd and even bits, the output of odd and even streams is governed by following rule.

Case	Past bit	Present bit	Output
1	0	0	No output
2	1	1	No output
3	0	1	$\frac{\sin x}{x}$ output for two bit durations
4	1	0	$\frac{\sin x}{x}$ phase reversed for two bit durations

The binary data is precoded as shown in Fig. 4.2. The data stream is split into even and odd streams using D flipflop. Using a set of JK flipflops we divide clock into CLK/2, CLK/4, 90^{o} phase shifted CLK/4. The CLK/4 and 90^{o} phase shifted CLK/4 is integrated to obtain triangle waveforms of four clock durations. These triangles are either positive going or negative going. If in each of odd or even streams we have present bit as 1 and pastbit as zero, we allow a positive triangle and if present bit is zero and past bit was one we allow negative triangle.

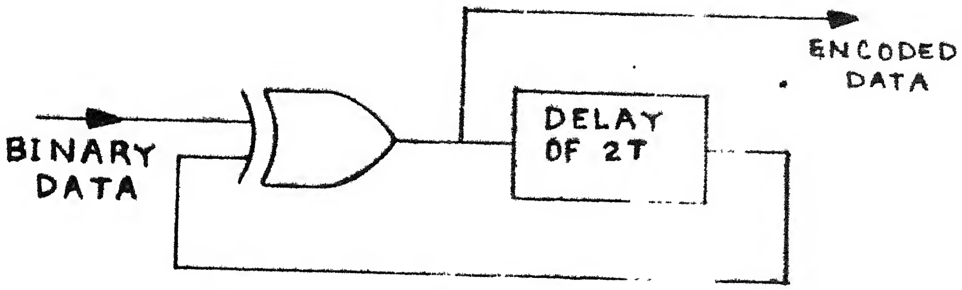


Fig 4.2 PRECODER FOR MDB

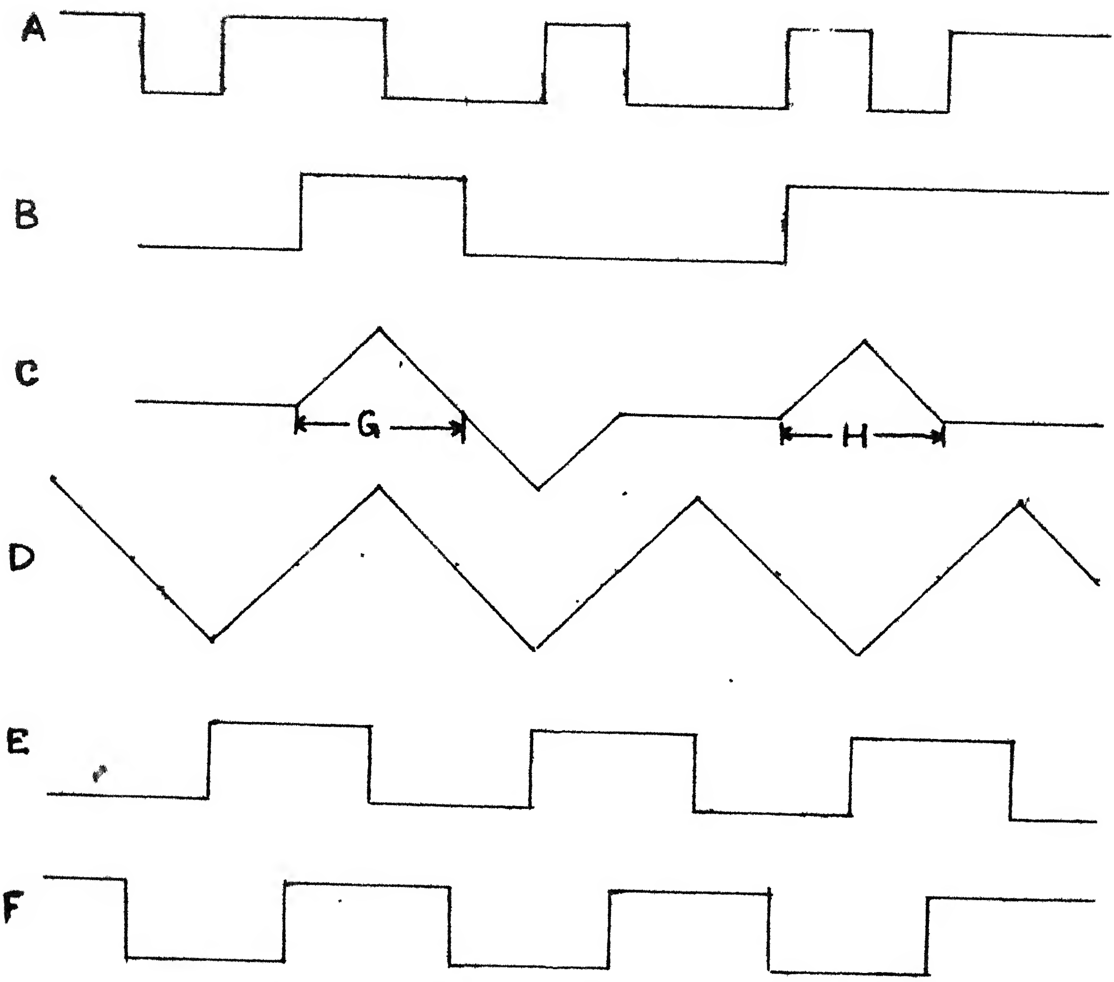


Fig 4.3 TIMING DIAGRAM FOR MDB (EVEN BITS)

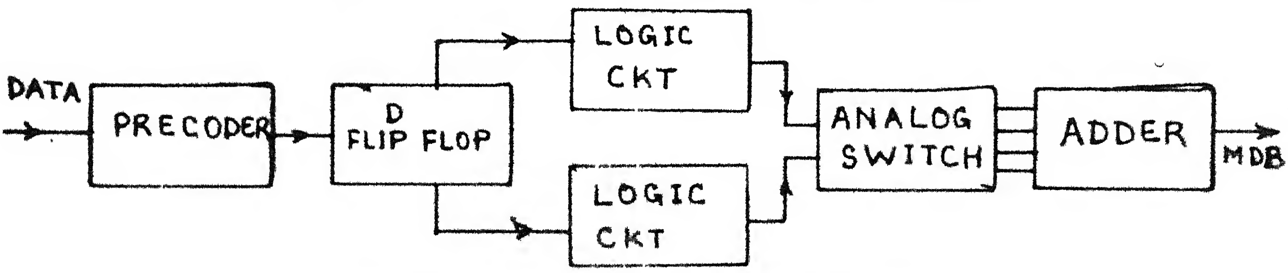


Fig 4.4 BLOCK DIAGRAM FOR MDB

These odd and even bits are then switched by analog switches using IC4066 and finally added using op-amp adder.

4.1.1 Timing Diagram of MDB:

Fig. 4.3 shows the timing diagram for even bit stream of data. A is the precoded data, B is even bit stream. The required output for this is shown in C. The output exists only if there was zero followed by one or one followed by zero. If there is zero followed by zero or one followed by one, there is no triangle output. Triangle shown in D is derived by integrating E.

Consider time slots G and H of Fig. 4.3. In both these a one follows a zero and in both cases a positive triangle is required at output. However a positive triangle exists during slot G but a negative triangle is available during slot H. So for same combination of zero-one the triangle should be uninverted during H. To achieve this we should have logic based on data (current and previous bit of data stream) and slot monitor waveform. Similarly odd bit stream is processed in the same way. Fig. 4.4 shows the block diagram of MDB realisation.

4.1.2 Logic Circuitory:

Four triangles are required to be switched by control logic Fig. 4.6. These triangles are obtained by integrating waveforms

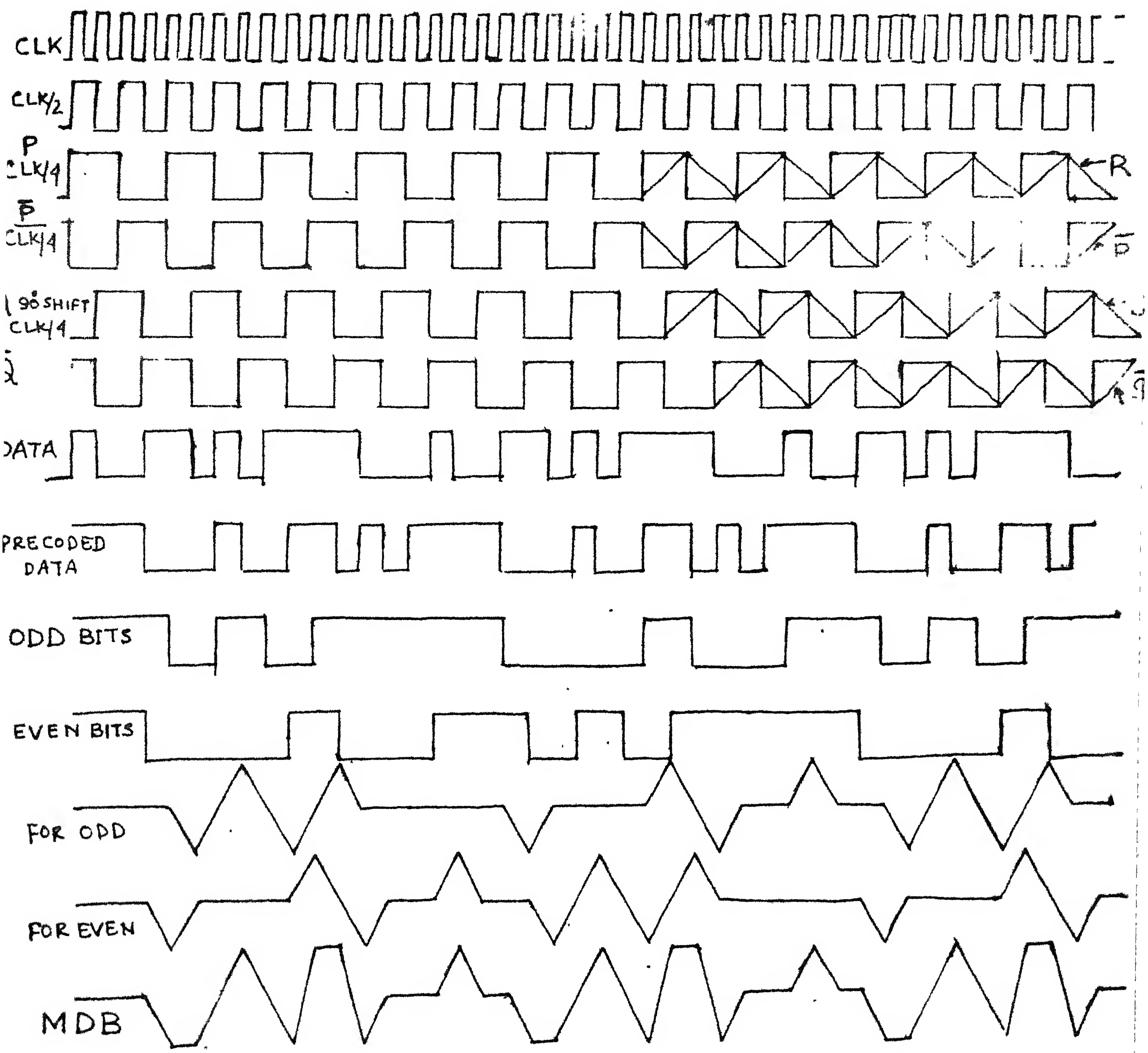


Fig 4.5 COMPLETE TIMING DIAGRAM

LOGIC :

$$R = A\bar{B}P + \bar{A}B\bar{P}$$
$$\bar{R} = A\bar{B}\bar{P} + \bar{A}BP$$
$$S = \bar{A}'B'Q + A'\bar{B}'\bar{Q}$$
$$\bar{S} = \bar{A}'B'\bar{Q} + A'\bar{B}'Q$$

A = CURRENT BIT
B = PAST BIT
A', B' for EVEN BITS

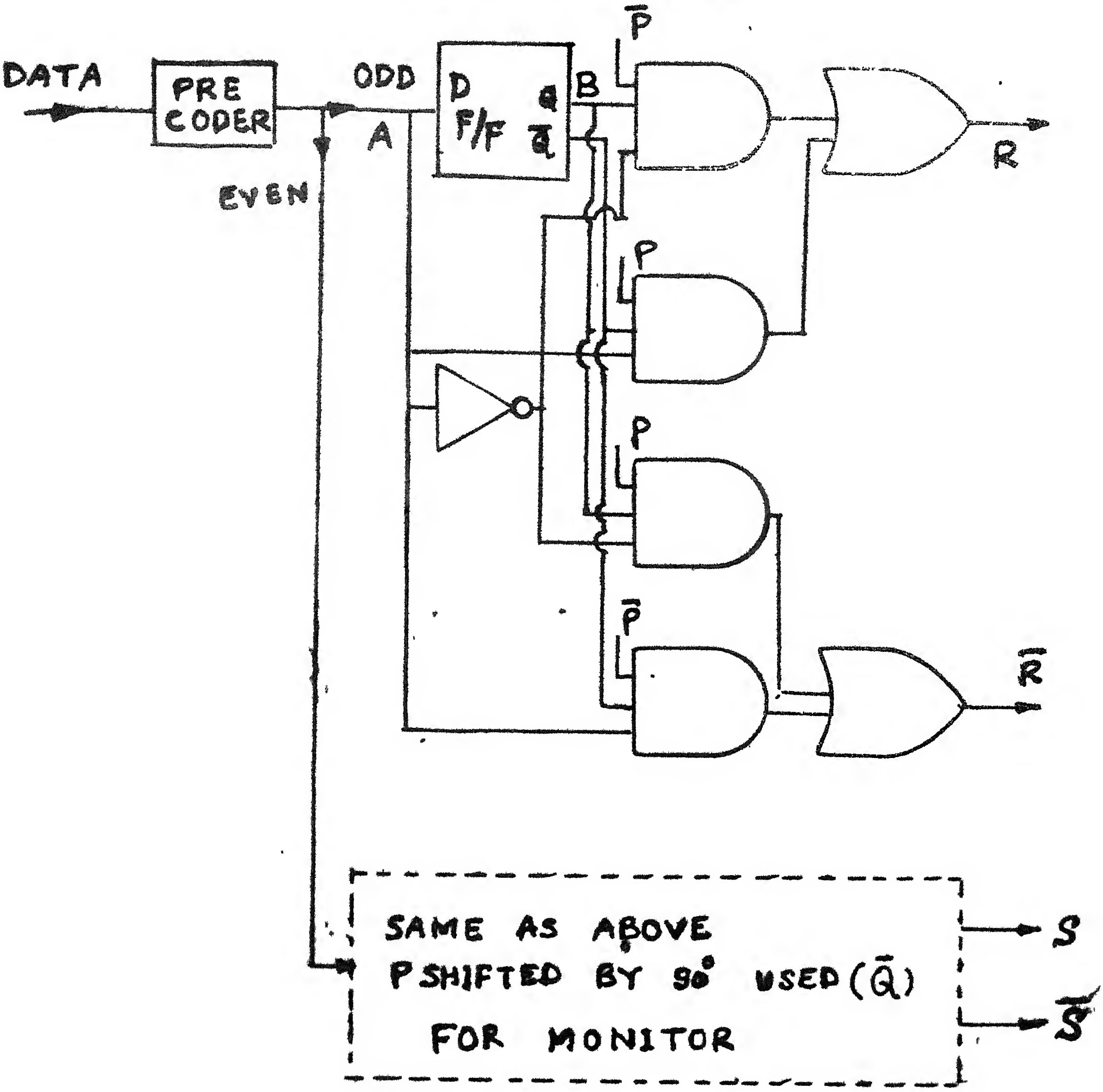


Fig 4·6 CONTROL LOGIC

of four clock duration i.e. divide clock by four and using JK flipflops get 90^{o} phase shifted CLK/4. By integrating and inverting these two waveforms we get four triangles i.e. triangle obtained by integrating clock of four clock durations (F), inverted version of this triangle, triangle obtained by integrating 90^{o} phase shifted clock of four clock durations (G) and its inverted versions. The complete timing diagram is shown in Fig. 4.5.

The control signals R, $\bar{R}$, S and $\bar{S}$ are used for switching on four triangles, available by integrating P and Q, through analog switches. The output of analog switches is fed to op-amp adder.

4.2 LOW PASS FILTER:

Before transmitting, the MDB is band limited by passing through a low pass filter. The filter used is shown in Fig. 4.7. Since we have assumed a data rate of 4.8K bits/sec we have designed the filter for cutoff frequency of 5 KHz. We use a third order Butterworth filter.

$$f_c = 5 \text{ KHz} \; ; \; w_c = 2\pi f_c = 2x3.142x5000$$

$$= 31428$$

$$30 \text{ K rad/sec.}$$

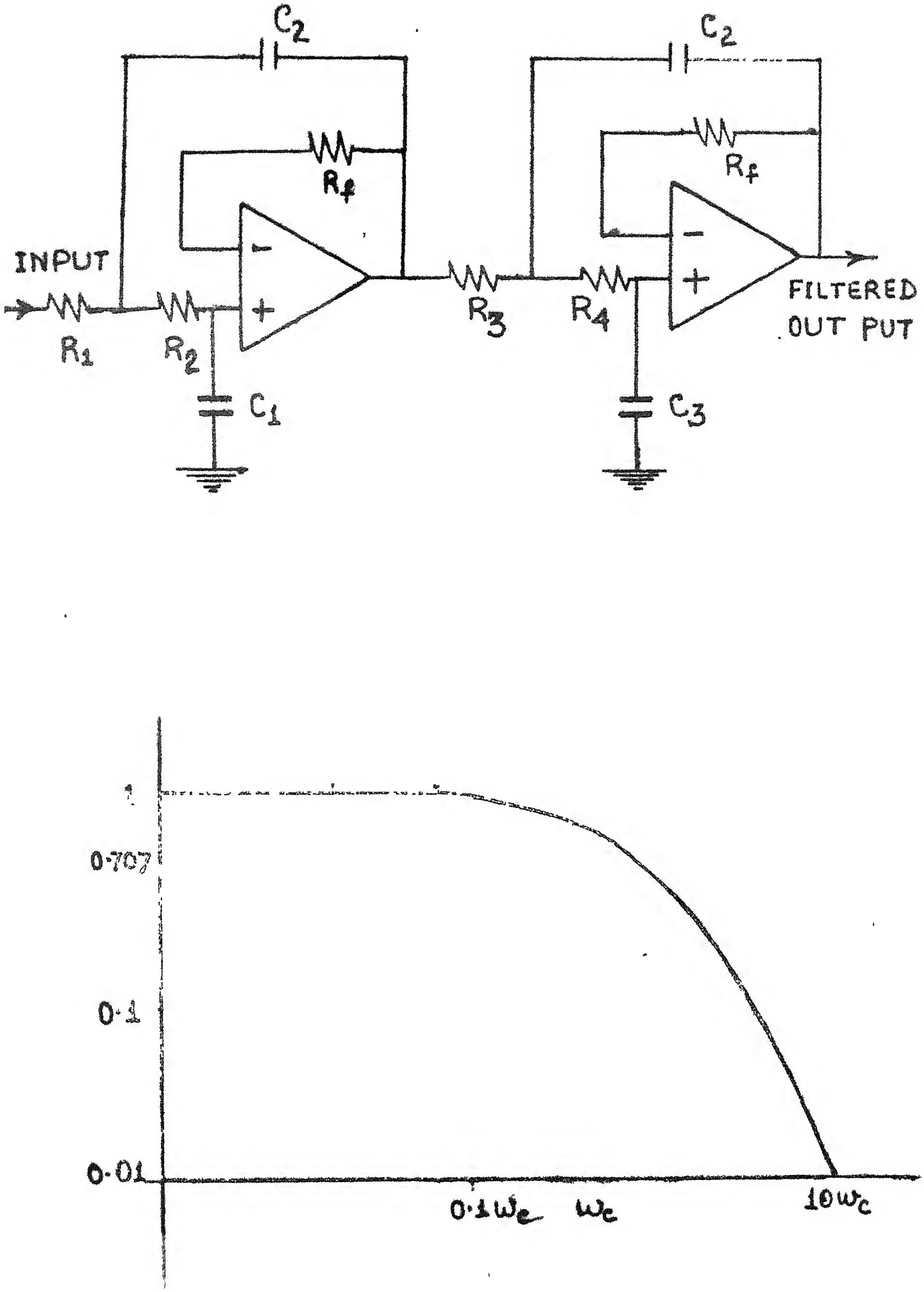


Fig 4·7 LOW PASS FILTER

Let $R_1 = R_2 = R = 10$ K ohm

$$C_1 = \frac{0.707}{w_c R} = \frac{0.707}{30x10^3 x10x10^3} = 0.0024 \ \mu Fd.$$

$$C_2 = 2C_1 = 2x0.0024 = 0.0048 \ \mu fd$$

4.3 HARDWARE IMPLEMENTATION, MDB AND TRANSMIT FILTER

The previous section described, how binary data is converted in to MDB form and then design of low pass filter before transmitting through radio. The actual hardware implementation is discussed in this section. Schematic diagram of MDB transmitter is shown in Appendix A, step wise implementation is explained below with the help of smaller circuits.

4.3.1 Precoder for MDB:

Precoding is done using two D flip-flops 74C175 and an exclusive OR 74C86, shown in Fig. 4.8(a).

4.3.2 Seperation of odd and even bits:

The precoded data is fed to a D flip-flop 74C175 using CLk/2 to get odd bits. To get even bits we use another 74C175 and trigger it with $\overline{CLK/2}$. It is shown in Fig. 4.8(b).

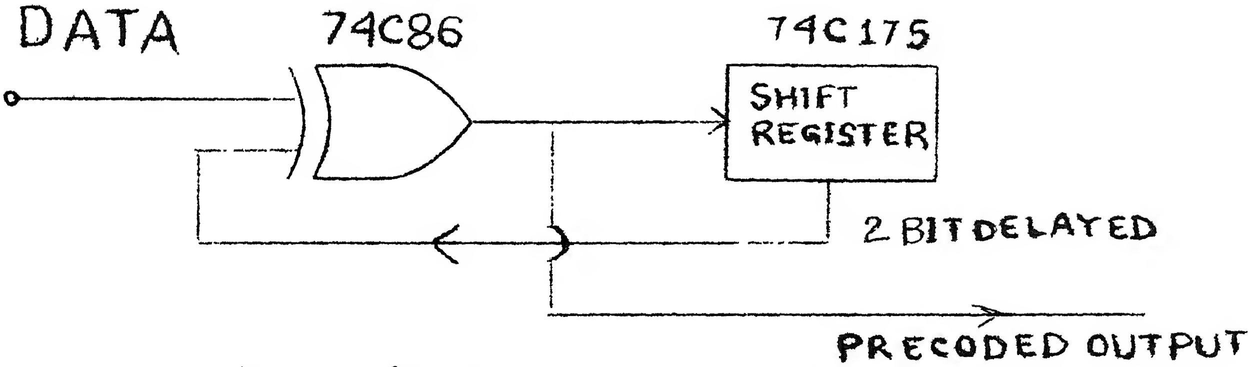


Fig 4.8(a) PRECODER FOR MDB

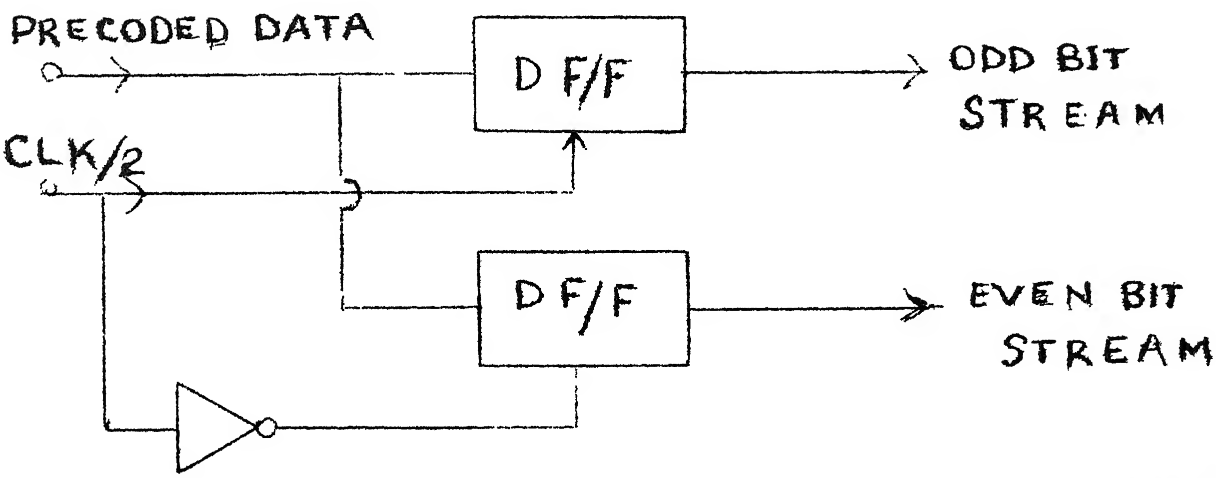


Fig 4.8(b) SEPERATION OF ODD & EVEN BITS

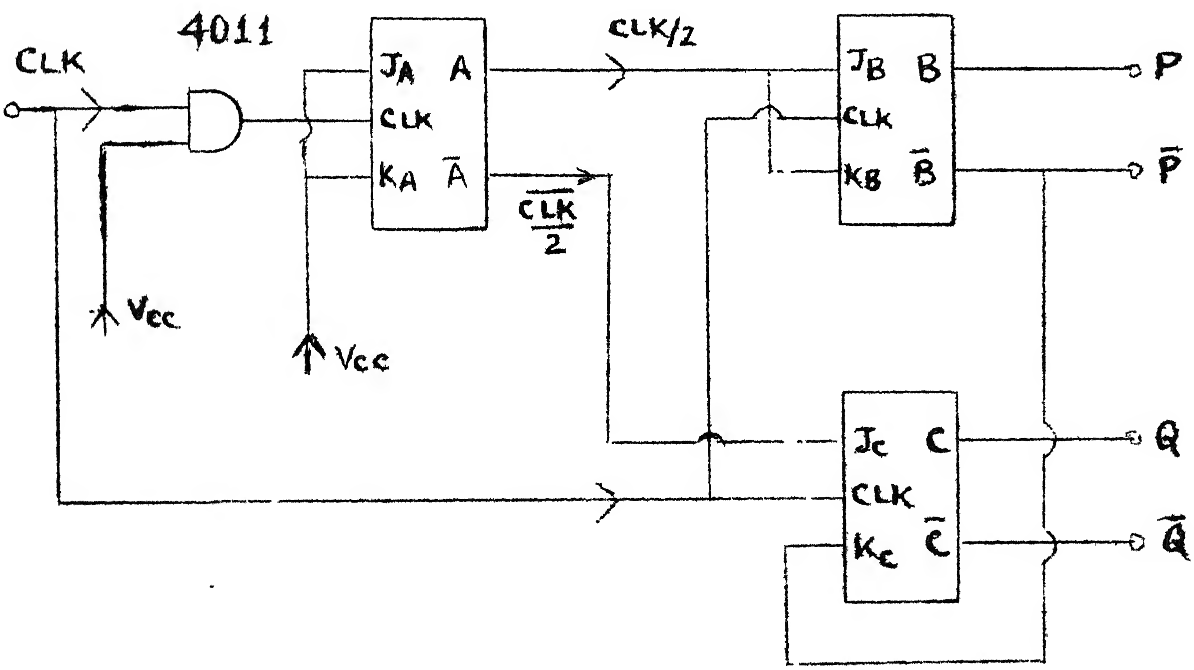


Fig 4.8(c) CLOCK DIVISION

4.3.3 Clock division:

The clock is fed through a buffer 4011 to a JK flip-flop 4027 to get clock of 2 durations and its inverted version. Fig. 4.8 (d) shows Karnaugh map to get clock of four durations and 90^{0} shifted clock of four durations and their inverted versions. Clock division is shown in Fig. 4.8(c).

4.3.4 Generation of Triangles:

Clock of four durations P is fed an op-amp integrator as shown in Fig. 4.8(e) to get triangle. This triangle is fed to inverter to get inverted triangle.

4.3.5 Control Logic:

The data is precoded by the circuit of Fig. 4.8(a) to avoid error propagation. The seperation of precoded data into odd and even bit streams is done by circuit of Fig. 4.8(b). The control signals which depend on odd and even streams are generated as explained in Fig. 4.6. These control signals are fed to four analog switches using IC 4066 to control the switching of four triangles generated above. The output of analog switches is added using simple adder giving MDB output as given in Fig. 4.8(g). The MDB output is fed to low pass filter having cutoff frequency of 5 KHz. The circuit diagram of filter is shown

	A	B	C
0	1	1	1
1	0	0	1
0	1	0	0
1	0	1	0
0	1	1	1

A \ BC	00	01	11	10
0	xx	1x	x1	1x
1	x1	xx	x1	xx

$J = K = 1$

A \ BC	00	01	11	10
0	xx	0x	xx	x0
1	1x	xx	x1	xx

$J_B = A$

$K_B = A$

A \ BC	00	01	11	10
0	xx	x1	xx	1x
1	x1	xx	x0	xx

$J_c = \bar{A}$

$K_c = \bar{B}$

Fig 4.8(d) Karnaugh Map for CLK Division

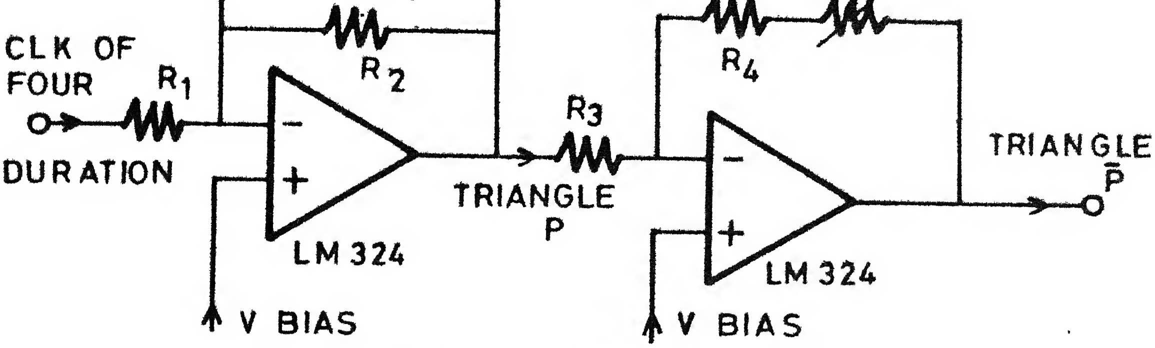


FIG. 4.8 (e) GENERATION OF TRIANGLES

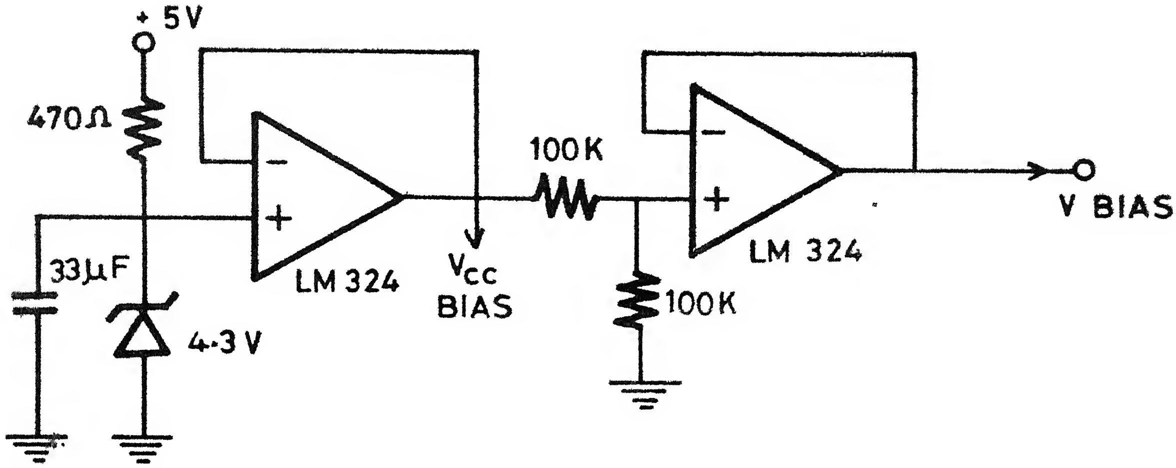


FIG. 4.8 (f) GENERATION OF V BIAS FOR LM 324

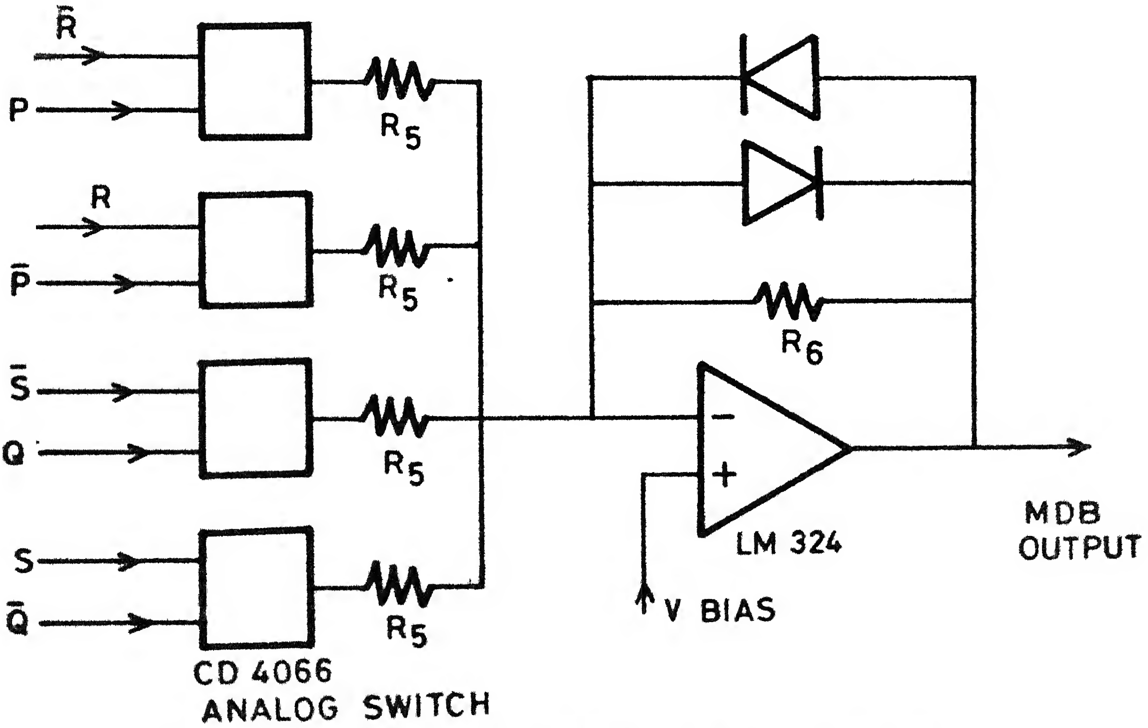


FIG. 4.8 (g) ANALOG SWITCH AND ADDER

in Fig. 4.7. The band limited output is fed to transmitter where it is frequency modulated. Fig. 4.5 shows the timing waveform to get MDB.

4.4 ERROR DETECTION IN MODIFIED DUOBINARY:

Modified duobinary has distinct advantage of error detection as compared to zero memory system. Error detection in zero memory system requires redundancy. MDB, however have finite memory and this can be utilized to monitor and detect errors [24] without introducing redundancy digits at transmitter. Distinctive patterns exist in the $1-D^2$ waveform. The modified duobinary has 3 levels, and the pulse train is divided into odd and even bits. Both odd and even pulse trains follow the same pattern. The rule for odd as well as for even bits is as follows. Two successive bits at the extreme levels always have opposite polarity.

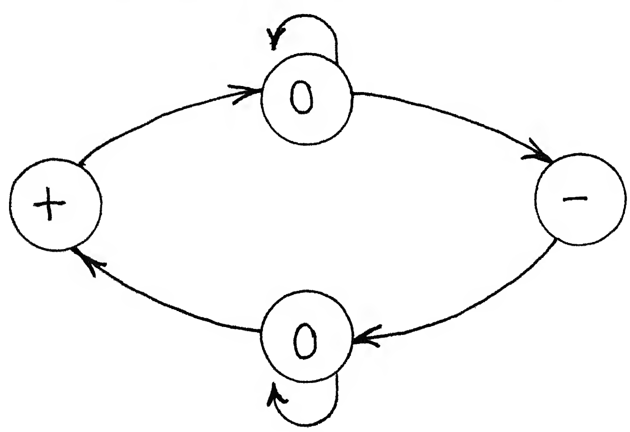
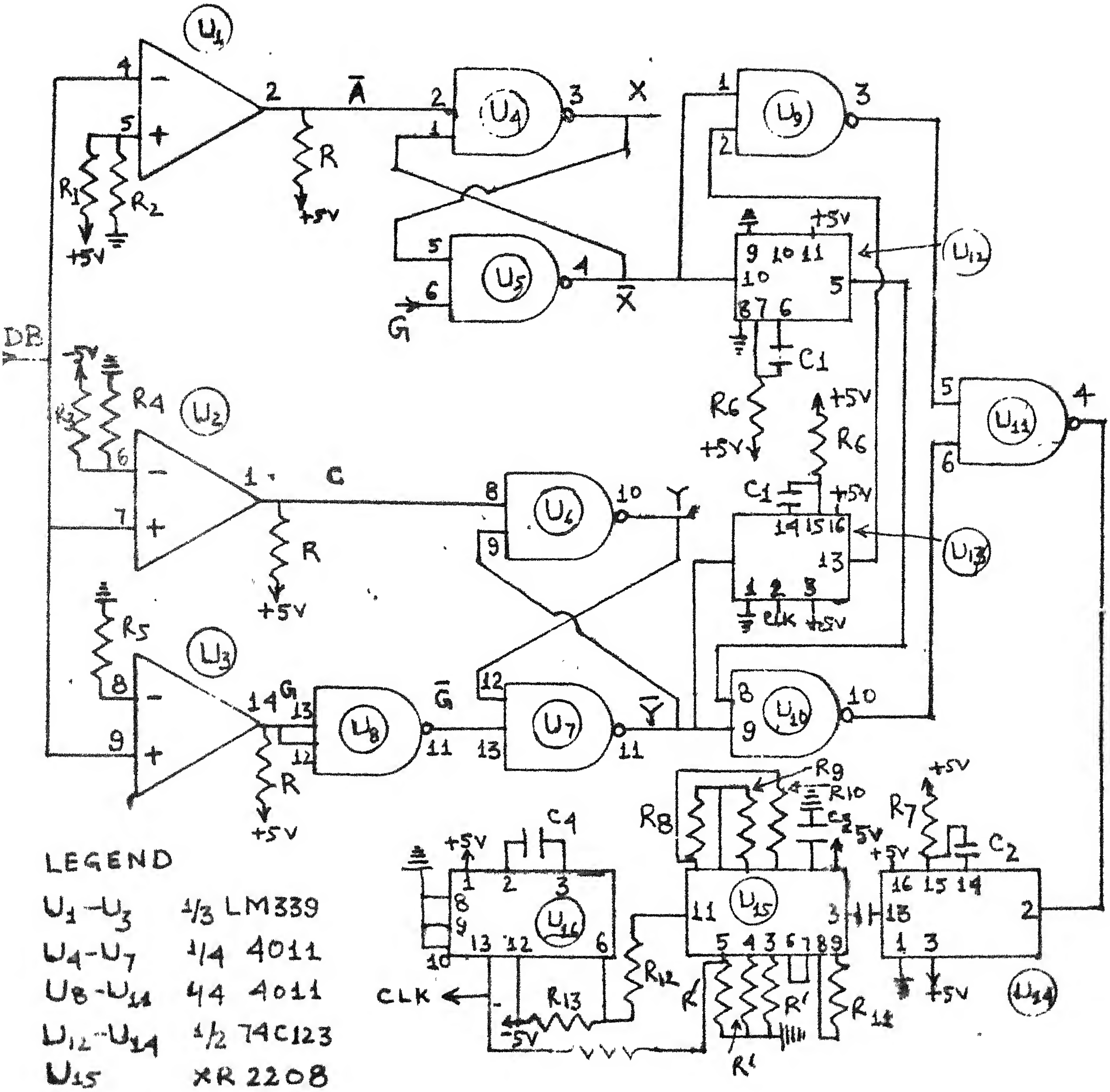


Fig. 4.9: State Diagram

A scheme for error detection has been given by Lender [24].

4.5 RECOVERY OF TIMING INFORMATION

Since packet radio will be used mainly for bursty traffic, it is essential that the receiver clock is synchronised with the transmitter clock within a period of time, which is short compared to a packet duration. Each station has a crystal which generates a clock of same frequency as the master clock. In order to synchronise local clock with the master clock, one scheme has been tried. In this the master sends an RF burst to mark the begining of the slot. Then the transmitting station sends a stream of 10101010 ... long enough, (approx. 70 bits) so that the receiver is able to synchronise with the transmitter. In one such scheme of clock recovery the clock transitions in MDB are made use of. These clock transitions are detected using two threshold comparators and a zero crossing detector as shown in Fig. 4.10. These clock transitions are then used to trigger a Monostable multivibrator which gives pulses of $T_c/4$ duration, where T_c is a clock period. The output of the Monostable is fed to a PLL at twice the clock frequency to recover the clock.



LEGEND

U_1–U_3	1/3 LM339
U_4–U_7	1/4 4011
U_8–U_{11}	44 4011
U_{12}–U_{14}	1/2 74C123
U_{15}	XR 2208
U_{16}	XR 2207

Fig 4.10 CLOCK RECOVERY

CHAPTER 5

CONCLUSION

In the present work a packet radio based data network has been studied. Various protocols applicable to packet radio environment have been reviewed and a suitable protocol has been suggested in Section 3.1.

Modified Duobinary signalling has been chosen for spectral reshaping of base band signal for data transmission over the radio link. Using MDB, collision detection is easier as MDB waveform has well defined state transitions which will be corrupted if there is a collision. Hardware implementation of MDB transmitter has been explained in Chapter 4 and its performance checked in terms of eye openings.

Use of MDB signal for transmission over FM radio will provide a packet radio network between various users, who may be mobile or scattered over a geographical area. An attempt has been made to make it compact and consume low power by using single plus 5 volt supply and CMOS logic.

5.1 SUGGESTIONS FOR FURTHER WORK

The following areas are suggested for further work.

a) An indepth study of packet radio protocol for finite no. of users as explained in Chapter 3 is required.

b) The system has been designed for a data rate of 4.8 K bit/sec on the assumption that a voice channel is available. Data rate can be increased if more channel band width is available.

c) Modified Duobinary has been used by Lender and others [27] for continuous data transmission. For such data clock recovery is achieved by nonlinear processing and narrow band PLL's. In some cases a seperate channel can be used for clock transmission. In case of packet radio, the data transmission is bursty in nature and of short duration. Hence there is a need for fast acquisition and synchronisation of receiver clock. A scheme for fast clock recovery has been described in Chapter 4, but there is a need to do more work in this direction.

d) An alternative scheme for clock recovery, which has not been tried is suggested in Fig. 4.11. The crystal clock of receiver is used to trigger voltage controlled multivibrator (VCM), The monostable pulses are used to sample MDB at maximum eye opening. The sample and hold output is fed to eye opening estimator which controls the voltage V_{in} of VCM. The VCM pulses at which eye opening for MDB waveform is maximum, will give us clock and hence data can be recovered out of MDB.

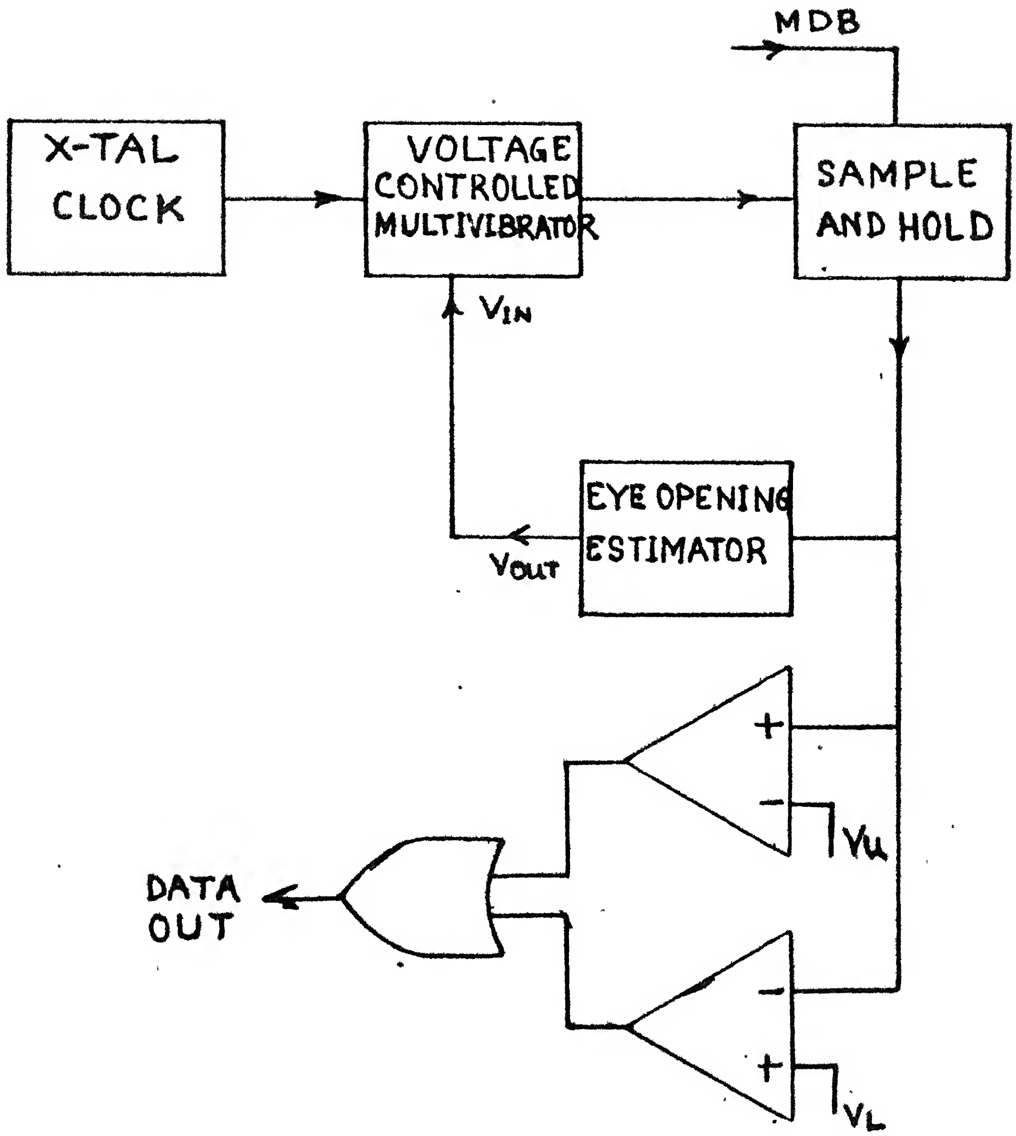
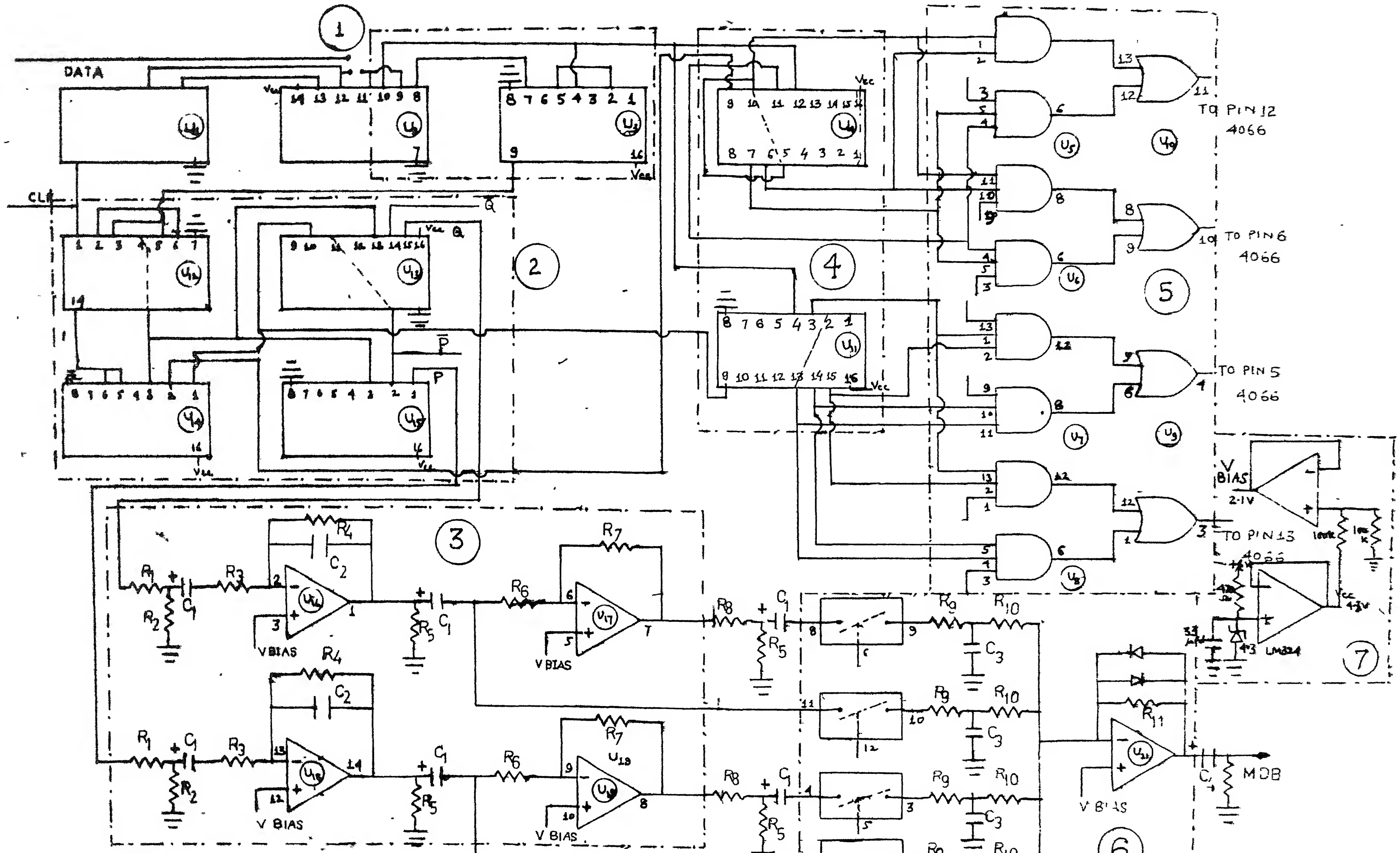


Fig 4.11 CLOCK RECOVERY USING VOLTAGE CONTROLLED MULTIVIBRATOR



LEGEND

U_1	IC 74C95	U_{12}	IC 4011	R_1	100 Ω	R_7	33 K	C_1	1.5 μFd		
U_2	IC 74C86	U_{13}–U_{15}	IC 4027	R_2	6.2 K	R_8	100 Ω	C_2	0.01 μFd		
U_3, U_4, U_{11}	IC 74C175	U_{16}–U_{19}	IC LM324	R_3	18 K	R_9	680 Ω	C_3	0.001 μFd		
U_5–U_8	IC 74HC11	U_{20}	IC 4066	R_4	100 K	R_{10}	51 K	C_4	0.2 μFd		
U_9 U_{10}	IC 4071	U_{21}	IC LM 324	R_5	62 K	R_{11}	58 K				
				R_6	33 K						

1. PRECODER
2. CLOCK DIVISION
3. TRIANGLE GENERATION
4. ODD & EVEN BITS
5. LOGIC CIRCUITORY
6. ANALOG GATE & ADDER
7. V-BIAS

SCHEMATIC DIAGRM MDB TRANSMITTER

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